Merlin PLUS
IP / ISDN / POTS Codec
User Manual

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1 Warnings & Safety Information

1. Both appliance power cables must be removed from the device for Power Disconnection.
2. Remove the phone cable from the POTS interface before servicing.

THUNDERSTORM AND LIGHTNING WARNING:

DO NOT USE Tieline codecs during thunderstorms and lightning. You may suffer an injury using a phone, Tieline codec, or any device connected to a phone during a thunderstorm. This can lead to personal injury and in extreme cases may be fatal. Protective devices can be fitted to the line, however, due to the extremely high voltages and energy levels involved in lightning strikes, these devices may not offer protection to the users, or 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 you are 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 remains connected to the phone or ISDN system, LAN 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.

DIGITAL PHONE SYSTEM WARNING:

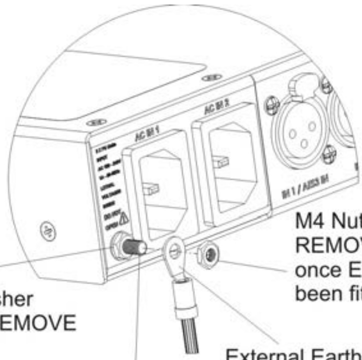
DO NOT CONNECT THE ANALOG POTS MODULE TO A DIGITAL PHONE SYSTEM. PERMANENT DAMAGE MAY OCCUR! If you are unfamiliar with any facility, check that the line you are using is NOT a digital line. If the Tieline codec becomes faulty due to the use of a digital phone system, the WARRANTY WILL BE VOID.

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 **NUT** 2 to connect your ground wire. The ground wire must have a suitable lug. When refitting **NUT** 2 ensure that both **NUT** 1 & **NUT** 2 are correctly tightened to establish and maintain a proper earth connection.



SAFE LISTENING GUIDANCE

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

Over exposure to excessive sound levels can damage your ears resulting in permanent noise-induced hearing loss (NIHL). Please use applicable health and safety authority guidelines on maximum exposure limits. As a rule of thumb, avoid extended periods listening to sound pressure levels (SPLs) of 85dBA or higher.

Warranty and Disclaimer

This equipment manufactured by Tieline is warranted by Tieline against defects in material and workmanship for two years from the date of original purchase. 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.

2 How to Use the Documentation

Manual Conventions

Warnings: Instructions that, if ignored, could result in death or serious personal injury caused by dangerous voltages or 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.

IP Information specific to IP connections.

ISDN Information specific to ISDN connections.

POTS Information specific to POTS connections.

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 *i* (information/help) button when navigating codec menus to display a dialog suggesting the actions which can be performed from within the current menu.

3 Glossary of Terms

AES/EBU	Digital audio standard used to carry digital audio signals between devices
AES3	Official term for the audio standard referred to often as AES/EBU
BRI	Basic Rate Interface for ISDN services
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.
DN	Directory Number for ISDN
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 is a field in an IP packet header for prioritizing data when traversing IP networks
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
ISDN	Integrated Services Digital Network
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
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
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.
MSN	Multiple Subscriber Number for ISDN
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.
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
PSTN	Public switched telephone network which is another term for POTS (see previous)

PSU	Power Supply Unit
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
RTP	A standardized packet format for sending audio and video data streams and ensures consistency in the delivery order of voice data packets
Runtime (edits)	Configuration changes which have not yet been saved, e.g. Matrix Editor edits.
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
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.
SPID	Service Profile ID for identifying devices over ISDN networks
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.
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 Getting to know Merlin PLUS

The 1RU Merlin PLUS rack mount codec is designed to rapidly expand your remote capability.

Merlin PLUS has the ability to manage up to 6 bidirectional mono connections, which saves money on codec hardware costs and reduces studio rack space requirements. Connect to IP codecs or smartphones using Report-IT, as well as ISDN and POTS codecs via optional plug-in transport modules. Merlin PLUS can also deliver high quality bidirectional stereo and full duplex communications for remote broadcast connections.

Overview of this User Manual

Use this manual to learn how to:

- Configure codec 'programs' (please read About Program Dialing for more info).
- Adjust audio and connection settings within the codec.

Please read Getting Connected Quickly for an overview of how to adjust and store audio and connection settings in your codec using 'programs'.

Applications

Merlin PLUS is ideal for studio and remote truck installations and is capable of:

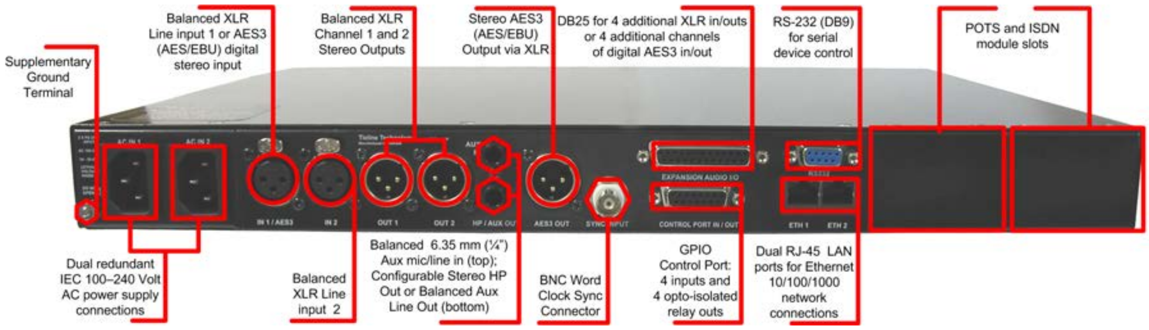
- 6 bidirectional mono remote connections.
- 3 bidirectional mono connections.
- 1 or 2 Bidirectional mono or stereo remotes, each with a separate bidirectional IFB channel for communications.
- 1 x stereo and 4 x mono bidirectional connections.
- Simple local or remote command and control.
- Recallable codec configurations via ‘programs’.

Codec Features

- DSP-based architecture designed for continuous operation.
- Dual Gigabit (10/100/1000) Ethernet ports with automatic switching for redundancy.
- Auto switching, dual redundant AC power supplies.
- Uncompressed PCM audio plus the low-delay, cascade resilient aptX® Enhanced algorithm (capable of up to 24bit, 48kHz audio sampling)
- Other popular algorithms including LC-AAC, HE-AAC v1 and v2, AAC-LD, AAC-ELD, AAC-ELDv2, Opus, MPEG-1 Layer II and III, Tieline Music and MusicPLUS, G.722 and G.711.
- Asymmetric encoding.
- SmartStream PLUS redundant streaming for high reliability over IP networks without Quality of Service.
- Fuse-IP bonding.
- Matrix Editor and Program Scheduler.
- IPv4 & IPv6 compatible and ready.
- Supports ISDN and POTS connections via optional interface modules.
- SNMP and integrated alarm management.
- HTML5 Toolbox GUI enables remote codec control over WANs.

- Low latency in-band RS-232 auxiliary data channel.
- Programmable software rules engine via a GUI for Control Port functions.
- Streamlined codec wizards and GUI for configuration and control.
- Support for multiple languages: English, Spanish, Portuguese and French.
- Connect to all Tieline IP codecs and Report-IT iOS/Android suite of Apps.

5 Merlin PLUS Rear Panel Connections



XLR Analog and AES3 Inputs

XLR **IN1/AES3** and **IN 2** are balanced line inputs.

Input 1 can also be used as an AES3 (AES/EBU) digital input. This input accepts both mono and stereo digital AES3 signals.

XLR Analog and AES3 Outputs

XLR **OUT 1** and **2** are balanced analog audio line outputs.

AES3 OUT is an AES/EBU digital audio output. Both the analog and digital outputs can be used simultaneously and the AES3 output can send both mono and stereo signals via the single XLR output.

DB-25 Input/Output Expansion Connector

A female DB25 connector can be used to attach a male DB25 Tascam (pin-out) breakout cable, providing 4 additional analog XLR in/outs, or 4 additional channels of digital AES3 audio in/out. Please note that expansion inputs/outputs 3, 4, 5 and 6 must be either all analog or all digital AES.

The additional audio inputs/outputs are grouped in pairs: inputs 3 and 4 and inputs 5 and 6. By default, if you program analog or digital audio in on expansion inputs then you will get the same output; i.e. analog in = analog out, or digital in = digital out. This can be adjusted if required. E.g. you can configure analog in on inputs 3, 4, 5 and 6 and AES3 digital out on outputs 3, 4, 5 and 6, or vice versa. Adjust the output setting independently if required via **SETTINGS** **> Audio > Output Type > [Select Outputs 3 to 6] > [Select output type].**

For analog sources use inputs 3, 4, 5 and 6. For digital AES3 sources use input 3 for channels 3 and 4 and use input 4 for channels 5 and 6. Similarly, for analog outs use outputs 3, 4, 5 and 6. For digital AES3 audio out use output 3 for channels 3 and 4 and use output 4 for channels 5 and 6.

DB25 pin outs are available in Appendix B.

Dual Gigabit Ethernet Ports

The codec features two Gigabit (10/100/1000) RJ-45 Ethernet ports for IP connections. By default, the codec assumes **ETH1** is the primary LAN connection and **ETH2** is the backup LAN connection when in use. If you are only using one Ethernet port, always use **ETH1**. Note: When connected to a 1Gigabit link the green Ethernet port LED is illuminated solid green. When attached to a 100Mb link, the green Ethernet port LED is not illuminated.

Aux Mic/Line Input

AUX IN 6.35mm (1/4") balanced auxiliary mic or line input.

Headphone Out/Aux Line Out

HP/AUX OUT 6.35mm (1/4") software configurable stereo headphone output, or balanced auxiliary line output. The front panel **HEADPHONE** output and rear panel **HP/AUX OUT** share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

Sync Input

BNC type **SYNC INPUT** for attaching Word Clock sync to the codec.

Command & Control Interfaces

1. Four relay inputs and four opto-isolated outputs for machine control via the DB15 **CONTROL PORT IN/OUT** connector.
2. A nine pin female **RS-232** serial connection for local and remote control of equipment at either end of the link.

Dual Redundant AC Power Inputs

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

Dual Module Slots

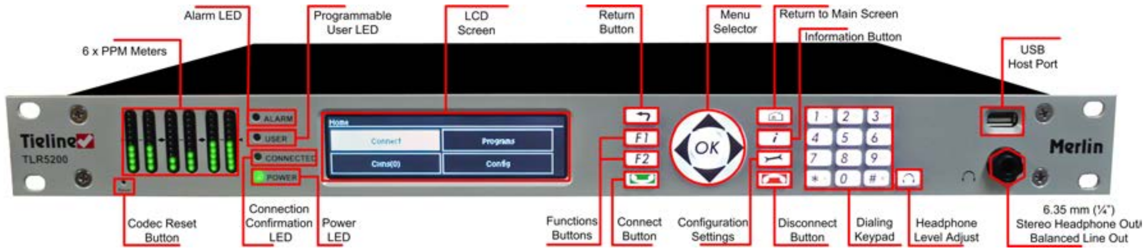
Two additional module slots for inserting optional POTS or ISDN modules.

Supplementary Ground Terminal

Supplementary ground terminal for connecting the unit to a ground connection. See [Warnings and Safety Information] for more details.

6 Merlin Front Panel Controls

The hardware front panel interface features menu navigation buttons, an LCD display with PPM metering and a dialing keypad.



Navigation Buttons

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

Dialing Keypad

The keypad has alpha-numeric buttons, plus star and hatch (pound) buttons, which can be used to enter contact and program information into the codec. Note: a virtual keyboard appears for entering alphanumeric characters in fields displayed on the codec **LCD SCREEN**.

Operation Button Descriptions

	Features	Operation Button Descriptions
	Return Button	Press to move back through menus & delete characters
F1	Function Button 1	Press to activate codec user functions and relays
F2	Function Button 2	Press to activate codec user functions
	Connect Button	Press to create an IP connection
	Home Button	Press to return to home screen
i	Information Button	Press to view a help menu onscreen
	Settings Button	Press to adjust codec settings
	Disconnect Button	Press to end a connection
	Headphone Button	Press to adjust headphone audio levels
Reset	Reset Button	Press to reboot the codec

Front Panel LED Descriptions

LED	LED Description
Alarm	The **ALARM LED** flashes red when an alarm is active in the codec. It stops flashing and illuminates solid red after an alarm has been acknowledged.
User (SIP)	The **USER LED** flashes orange when registering to an active SIP server account. It illuminates solid orange when the codec has been registered to an active SIP account successfully. Note: the LED flashes until all accounts have been registered successfully when multiple SIP accounts have been added.
Connected	When dialing a connection the **CONNECTED LED** flashes until the codec connects. Note: It also flashes until all streams / transports are connected when multiple streams / transports are connecting.
Power	The **POWER LED** indicates power is connected to the codec.

Stereo RTS Headphone Output

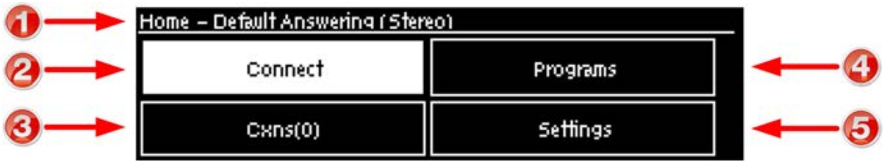
The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for audio monitoring and this can also be switched to a balanced mono auxiliary line output. The front panel **HEADPHONE** output and rear panel **HP/AUX OUT** share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

USB 2.0 Host Port

The USB 2.0 host port can be used for playback of backup audio files on a USB stick, firmware upgrades, or to attach a USB Wi-Fi dongle to the front panel of the codec.

7 Navigating Menus

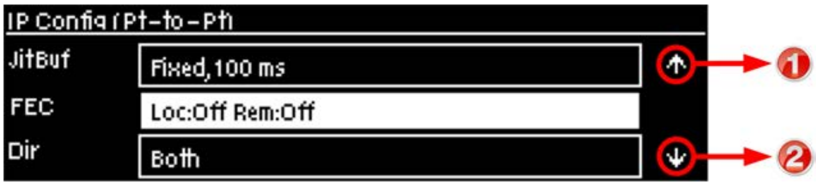
All main codec menus can be launched from the **Home** screen which includes:



	Features	Codec Home Screen Elements
1	Screen Name	The name of the current screen
2	Connect	Select to connect and adjust connection settings
3	Cxns	Displays the number of current connections and connection details
4	Programs	View and edit Program configurations
5	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.

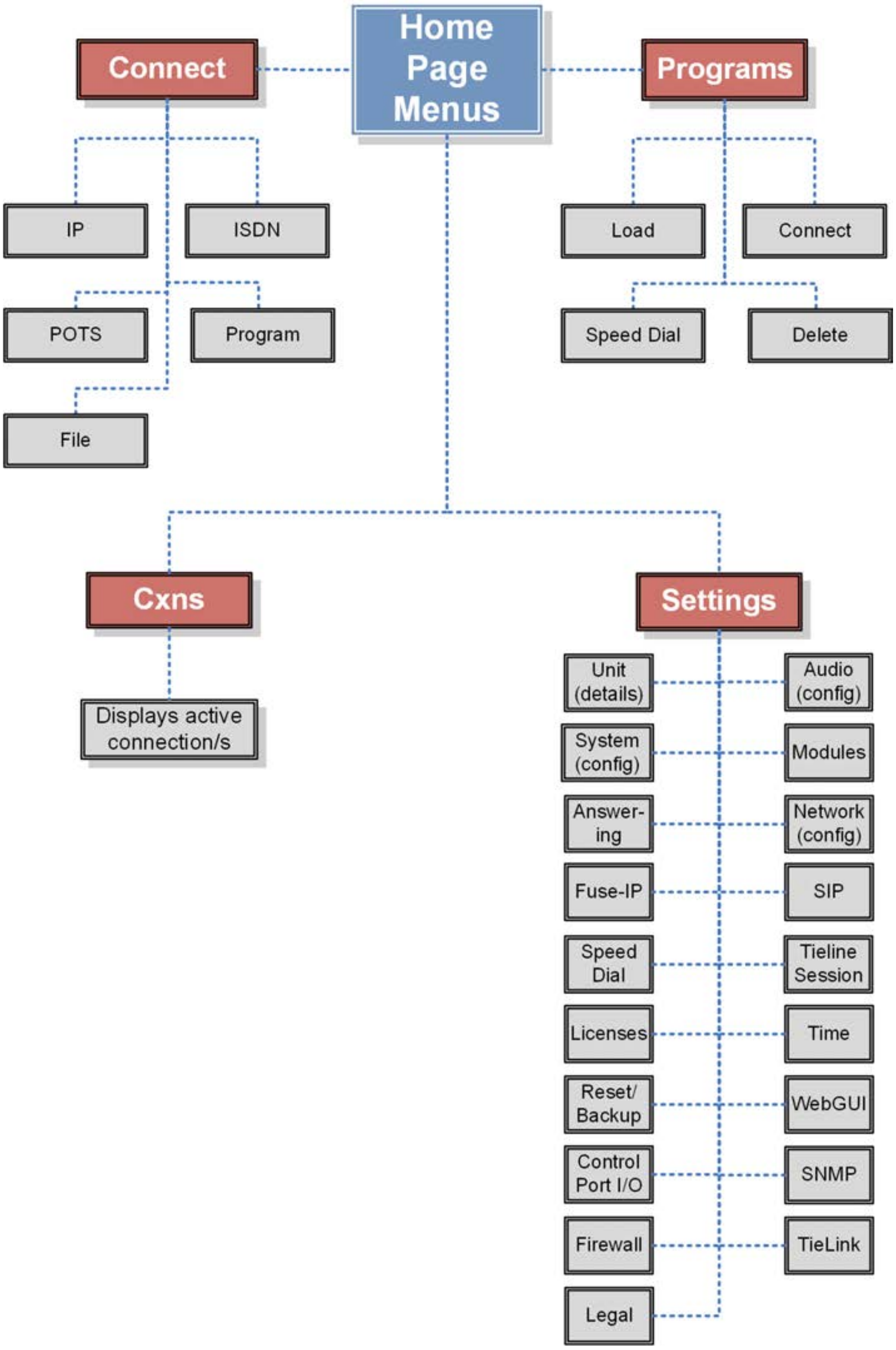
If a complete menu cannot be viewed on a single codec screen, arrows on the right hand side of the screen indicate that the current menu has options below and/or above the visible items. Use the navigation arrows to scroll up and down.



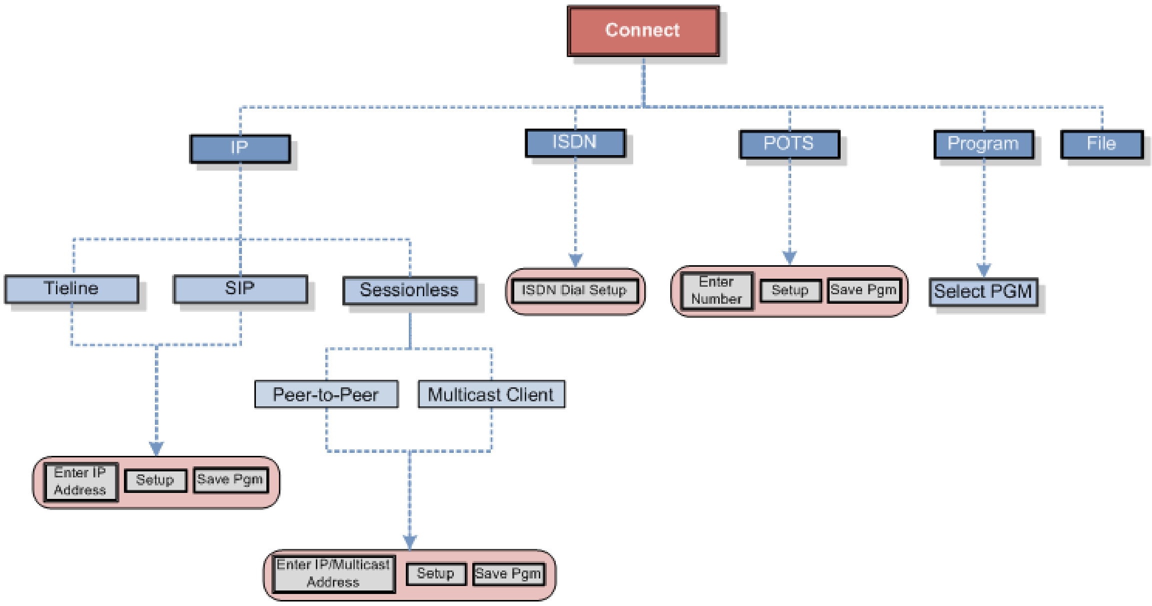
	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

Codec Menu Overview

Following is an overview of the codec menus from the **Home** screen. Note: file playback may not be supported in all codecs.

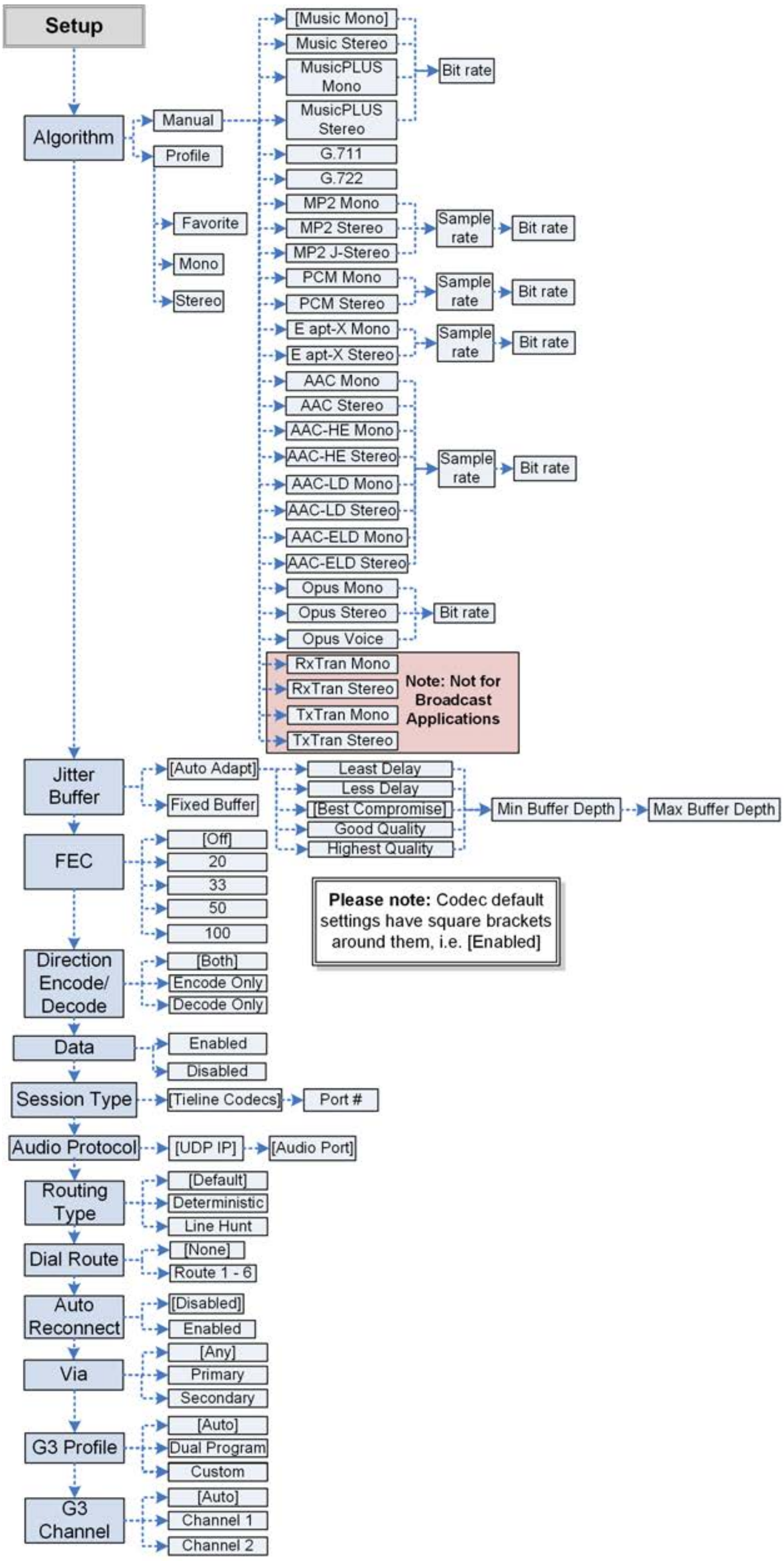


Connect Menu



IP Setup Menu Navigation

After selecting **IP** and a connection mode use **Setup** to adjust connection settings.

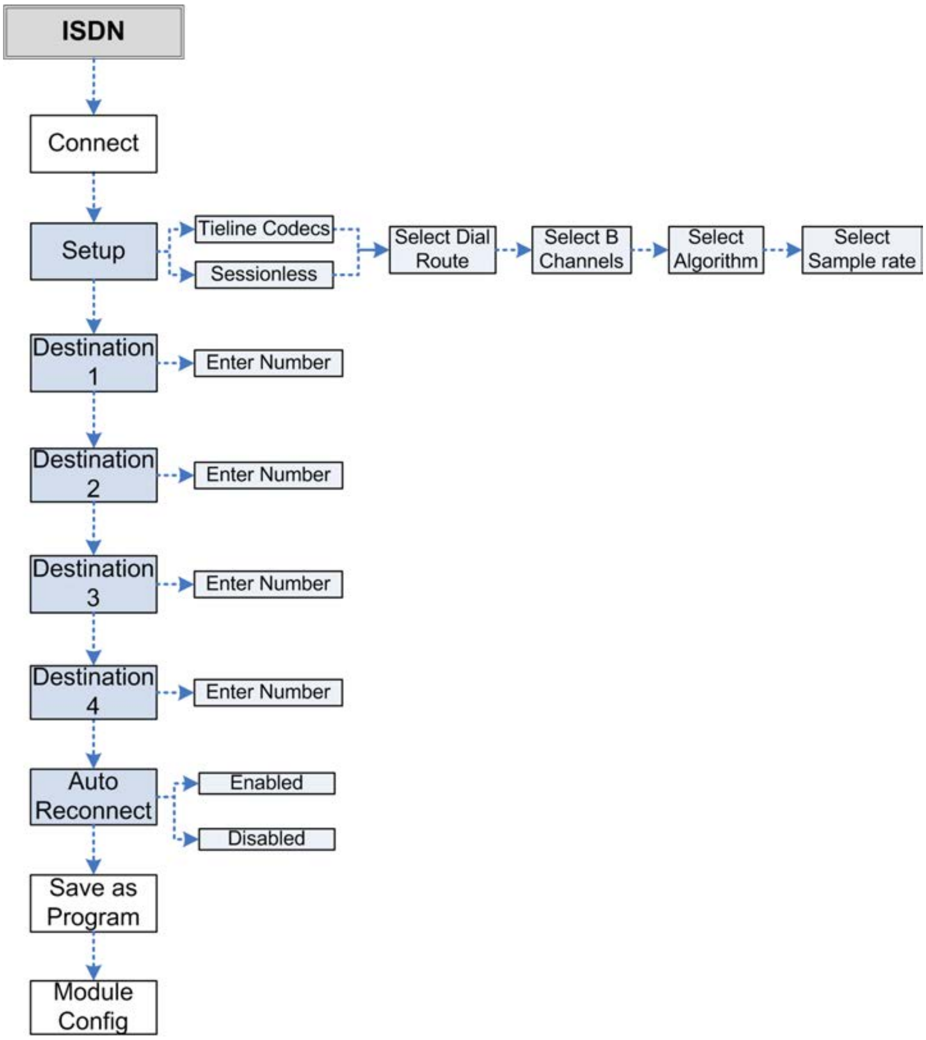


Important Notes:

- Depending on the session type selected in the codec, not all options are displayed. E.g. **Session Type** and **Data** are not displayed when configuring **Sessionless** IP connections.
- Default settings may also change depending on the session type selected, e.g. **Tieline Session** versus **SIP** or **Sessionless**.

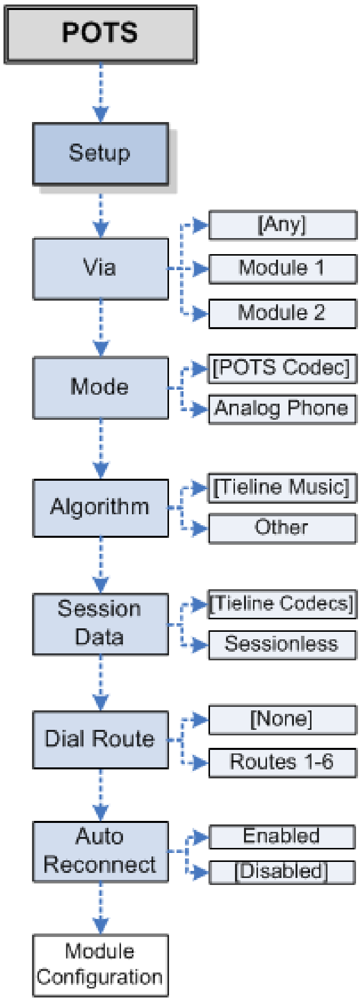
ISDN Menu Navigation

Select **Connect > ISDN** to configure ISDN dialing settings using the codec front panel.



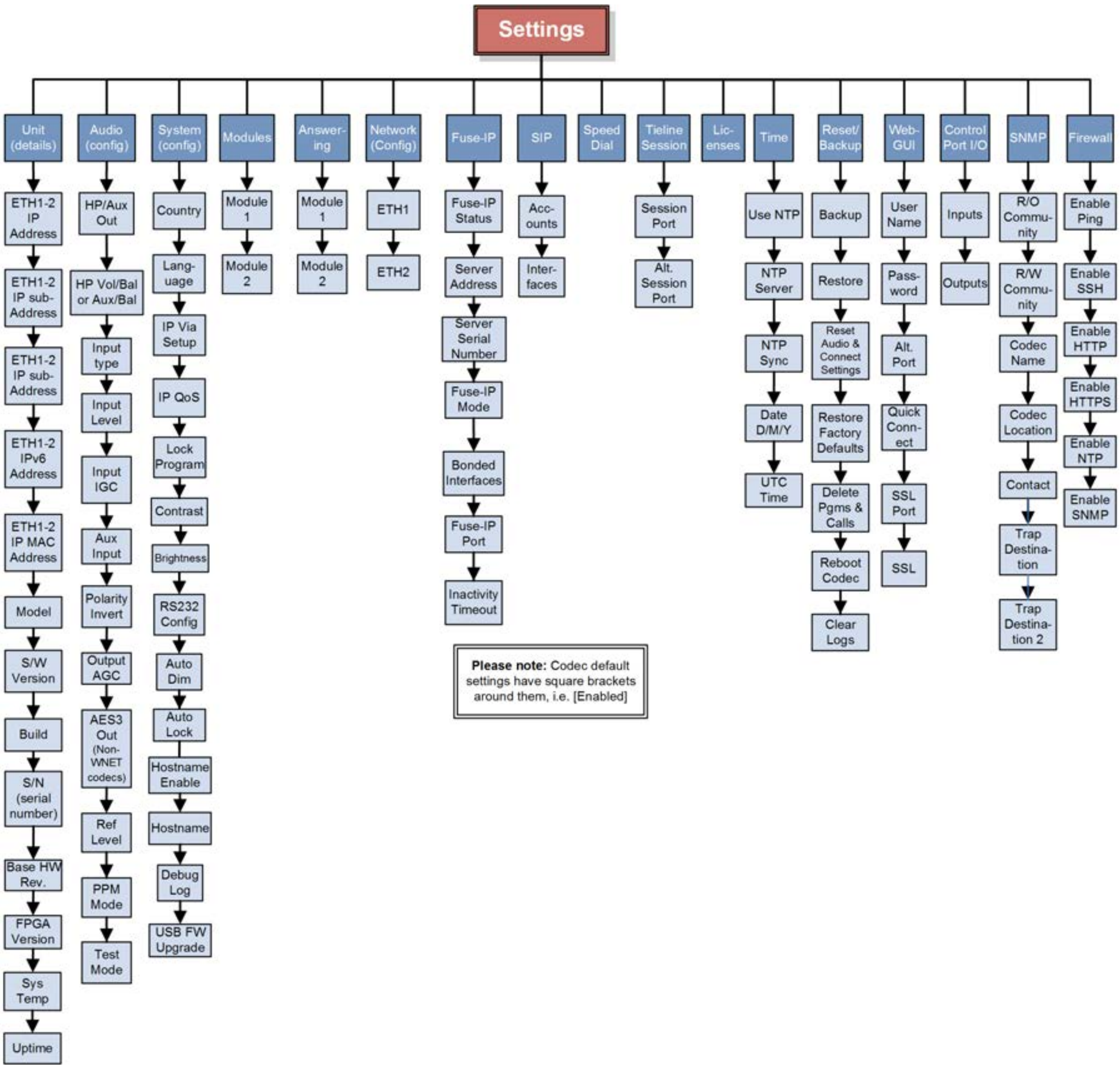
POTS Menu Navigation

Select **Connect > POTS** to configure POTS dialing settings using the codec front panel. Note: default settings are surrounded by square brackets.



Settings Menu

Press the **SETTINGS** button on the codec front panel to access a wide range of configuration settings.



8 Merlin PLUS Input Levels and PPMs

Important Note:

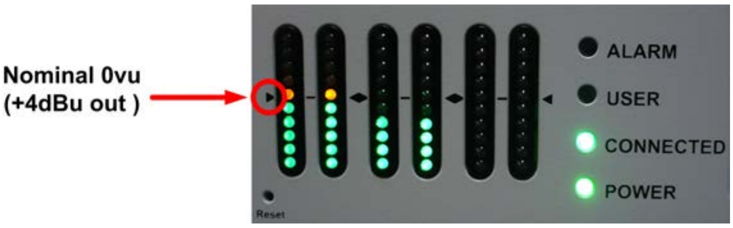
- See Configuring AES3 input audio for more information about digital in/outs. Input audio functions can also be configured using the Toolbox Web-GUI. See Configuring Input/Output Settings for more information.
- Merlin PLUS supports input expansion with inputs/outputs 3, 4, 5 and 6. Expansion inputs 3 to 6 must be either all analog or all digital. Expansion outputs 3 to 6 must also be either all analog or all digital. By default the output selection follows the input selection, however, inputs and outputs can be configured differently. For example, if you configure expansion inputs 3 to 6 as analog, you can configure all expansion outputs 3 to 6 as digital AES3. Adjust the output setting independently if required via **SETTINGS** **> Audio > Output Type > [Select Outputs] > [Select Analog or Digital]**

Adjusting PPM Meter Reference Scale Settings

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 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 G5 audio reference scale displayed on the PPMs when you connect to a Tieline G5 codec is -38dBFS to 0dBFS (e.g. Merlin, Genie, Bridge-IT and ViA codec families). Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 18dB of headroom from the nominal 0vu point. The comparison table below outlines the reference scales for G5 and G3 codecs and Report-IT in dBFS, as well as the equivalent dBU scale.



The audio reference level settings in the codec are:

	Reference Level	Description	dBu	dBFS
1	Tieline G5	PPM meter low point	-16dBu	-38dBFS
		Nominal 0vu reference level	+4dBu	-18dBFS
		Level at which audio will clip/distort	+22dBu	0dBFS
2	Tieline G3	PPM meter low point	-11dBu	-29dBFS
		Nominal 0vu reference level	+4dBu	-14dBFS
		Level at which audio will clip/distort	+18dBu	0dBFS
3	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 G3 Codec Audio Reference Scales

Genie, Merlin, Bridge-IT and ViA 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.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Ref Level** and press .
4. Select **Tieline G3** and press .

Configure Report-IT Audio Reference Scales

The **Report-IT** setting is used for compatibility when streaming audio using Tieline's Report-IT smartphone application.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Ref Level** and press .
4. Select **Report-IT** and press .

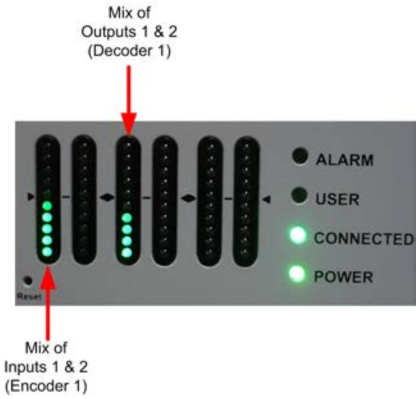
The Report-IT metering scale is between -23dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -10dBFS. Audio peaks should not exceed 0dBFS.

Default PPM Metering Displayed

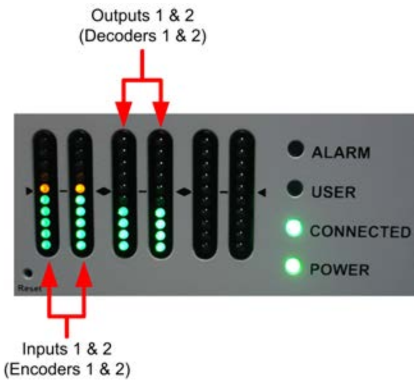
The following PPM settings are displayed by default in the codec. These settings change after making matrix routing adjustments in the **Matrix Editor panel**. The **PPM Mode** can be adjusted via **Settings > Audio > PPM Mode**.

Mono and Stereo Metering

When connected with a mono program the codec will display a mix of inputs 1 and 2 on **PPM1**. **PPM 3** displays the level of return audio.



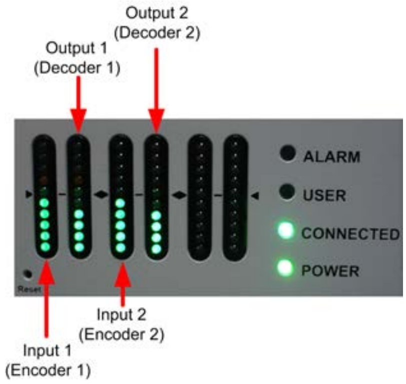
When connecting with a stereo program, the codec displays audio on **PPM1 & 2** for inputs 1 and 2 and **PPM 3 & 4** for the return program audio.



Important Note: When sending an audio stream to multiple endpoints using a multi-unicast program, the codec PPMs will display the same audio monitoring as for a default mono or stereo connection. The return audio PPM metering will display return audio from the first connection dialed within the multi-unicast program.

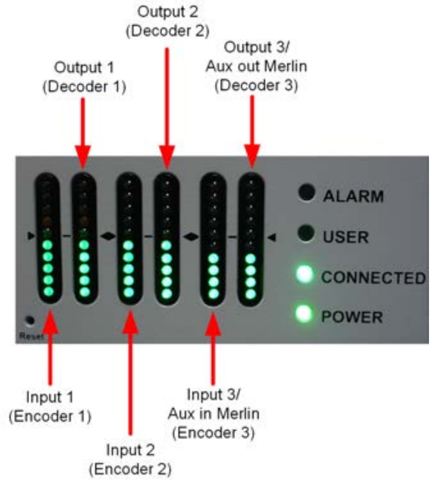
2 x Mono Peer-to-Peer Connection Metering

The codec is capable of creating two independent mono audio stream connections simultaneously. In this situation the codec will display outgoing connection 1 audio on **PPM1** for input 1 and use **PPM 2** to display return audio. Outgoing connection 2 audio will be displayed on **PPM3** for input 2 and **PPM 4** displays the return audio.



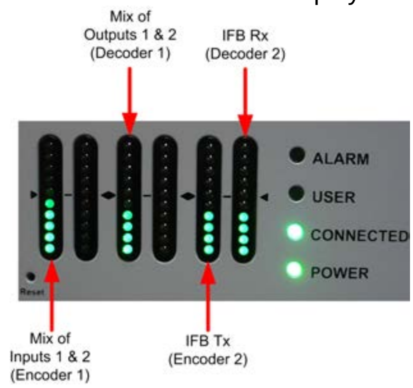
3 x Mono Peer-to-Peer Connection Metering

The codec is capable of creating three independent mono audio stream connections simultaneously. By default, the codec displays outgoing connection 1 audio on **PPM1** for input 1 and uses **PPM 2** to display return audio. Outgoing connection 2 audio is displayed on **PPM3** for input 2 and **PPM 4** displays the return audio. Outgoing connection 3 audio will be displayed on **PPM5** for input 3 and **PPM 6** displays the return audio.



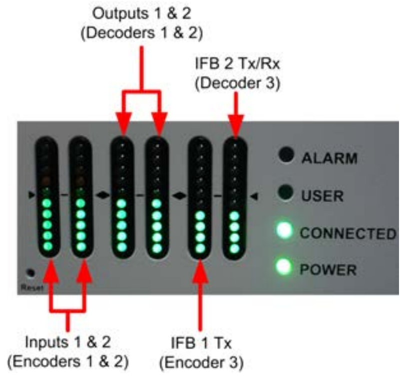
1 x Mono Peer-to-Peer + IFB Metering

This program transmits a bidirectional mono audio stream and a separate bidirectional mono IFB communications audio stream. A mix of inputs 1 and 2 is displayed on **PPM1**. **PPM 3** displays the level of return audio. **PPM5** and **PPM6** display IFB audio in/out.



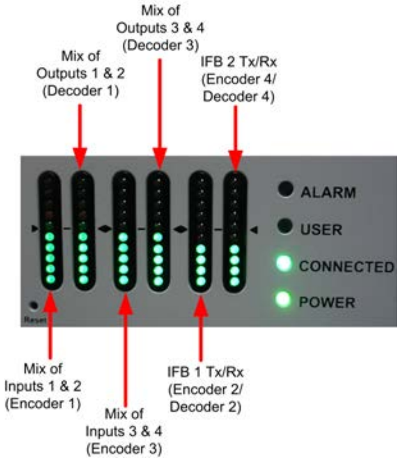
1 x Stereo Peer-to-Peer + IFB Metering

This program transmits a bidirectional stereo audio stream and a separate bidirectional mono IFB communications audio stream. Incoming and outgoing audio for both audio streams is displayed on the PPMs.



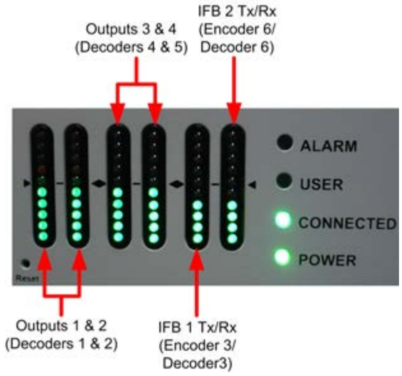
2 x Mono Peer-to-Peer + IFB Metering

This program transmits two bidirectional mono audio streams and two bidirectional mono IFB audio streams. Incoming and outgoing audio for both audio streams is displayed on the PPMs. A mix of inputs 1 and 2 is displayed on **PPM1** and a mix of inputs 3 and 4 is displayed on **PPM3**. **PPMs 2 & 4** display each return audio stream. **PPM5** and **PPM6** display a mix of IFB audio in/out for each IFB audio stream.



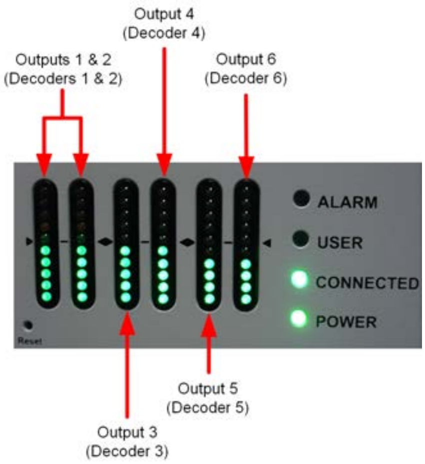
2 x Stereo Peer-to-Peer + IFB Metering

This program transmits two bidirectional stereo audio streams and two bidirectional mono IFB audio streams. Decoder audio for both stereo audio streams is displayed on **PPMs 1-4**. **PPM5** and **PPM6** display a mix of IFB audio in/out for each IFB audio stream.



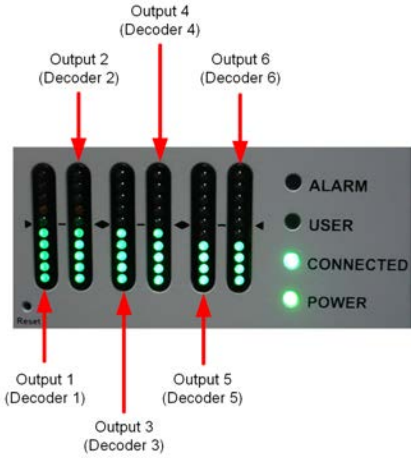
Metering Stereo and 4 Mono Peer-to-Peer Audio Streams

By default the PPMs display decoder audio levels when a **Stereo + 4 x Mono Peer-to-Peer** program is configured. Decoders 1-6 are mapped to **PPMs 1-6**, with decoders 1 and 2 for the stereo connection, followed by the 4 mono connections represented by **PPMs 3-6**. Note: All encoders and decoders can be monitored simultaneously using the PPMs panel in the Toolbox Web-GUI.



Metering 6 Bidirectional Peer-to-Peer Mono Audio Streams

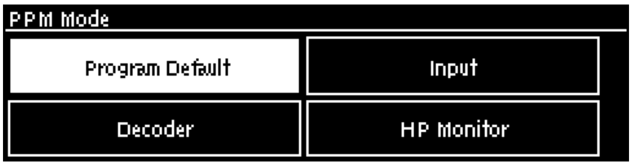
By default the PPMs display decoder audio levels when up to 6 mono audio streams are configured within a program. Decoders 1-6 are mapped to **PPMs 1-6**. Note: All encoders and decoders can be monitored simultaneously using the PPMs panel in the Toolbox Web-GUI.



Adjusting Default PPM Metering

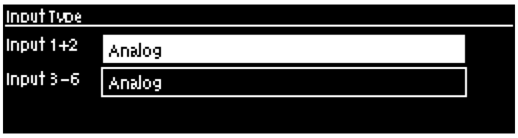
The default PPM metering settings can be adjusted via **Settings > Audio > PPM Mode**. The options include:

	PPM Mode	Description
1	Program Default (default)	Displays default program PPM meter settings (i.e. the settings described previously for mono, stereo programs etc.).
2	Input	Maps input encoders 1 to 6 with PPM meters 1 to 6.
3	Decoder	Maps decoders 1 to 6 with PPM meters 1 to 6.
4	HP Monitor	Maps PPM meters to inputs/outputs currently selected via the headphone monitoring function. The default headphone monitoring setting is accessed via **HEADPHONE** **> Monitor Source > [Select audio Source]**.

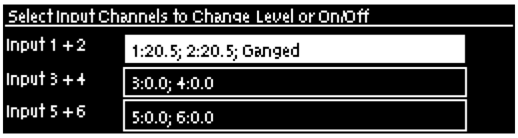


Selecting Analog and Digital Inputs and Adjusting Input Levels

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Inputs 1 and 2 and inputs 3 to 6 are displayed under **Input Type** and should be set to **Analog**; press to toggle between **Analog** and **AES3** and press the **RETURN** button to exit the menu.



4. Use the down navigation button to highlight **Input Level** and press the button.



5. Navigate to the channels you want to adjust and press .
6. Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press on the numeric keypad to toggle channel 1 on and off.
7. Use the left or right navigation buttons to select the appropriate gain setting, then press the button to save the settings.

Important Note:
- Gain adjustments can be made in 0.5dB increments.
- There is a maximum of 6dB of additional gain available when adjusting a digital input.
- To adjust levels quickly press and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
- 15 volt phantom power can only be supplied on the auxiliary input; this is disabled by default.

	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

Quick Level Adjustment of Input 1 and 2

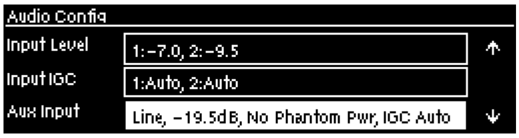
1. Press and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
2. Press on the numeric keypad to toggle channel 1 on and off and press to toggle channel 2 on and off.
3. Use the up and down arrow buttons to navigate to the channel you want to adjust. Note: A channel is highlighted when selected.

4. Use the left and right arrow buttons to adjust the input levels up or down.
5. Press the **RETURN** button to exit the screen.

Auxiliary Input Adjustment

The codec has 1 x 6.35mm (1/4") Mic/Line level Jack on the rear panel. By default the input is **Off** and can be configured by:

1. Selecting the **SETTINGS** button.
2. Navigate to **Audio** and press the button**.**
3. Use the arrow-down button to select **Aux Input** and press the button to view menu options.



Input settings which can be adjusted include:

- Input on/off.
- Input level.
- Input Type: High Gain Mic, Medium Gain Mic, Low Gain Mic, Unbalanced and Line Level.
- Phantom power (15V available when enabled).
- IGC.

Important Note: When the auxiliary input (**AUX IN**) is **On** the default mixer configuration sends audio to all inputs. If you are not using the auxiliary input ensure it is **Off** to avoid additional noise in program audio.

Ganging Audio Channels

It is possible to gang channels together and adjust the audio level of the ganged channels simultaneously. When channels are ganged together:

- Both channels highlight together when selected.
- The gain setting for both channels is automatically set to match the gain level of the lowest of the two channels when ganging is first configured.
- If one channel is turned on when ganging is first configured then the other one will be turned on automatically.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Use the down navigation button to highlight **Input Level** and press the button.
4. Navigate to the channels you want to gang and press the button.
5. Navigate to the **Gang** function and press the button to toggle between **Enabled** or **Disabled.**
6. Use the up and down arrow buttons to highlight and select the audio channels.
7. Use the left and right arrow buttons to adjust the levels for both inputs up or down simultaneously.
8. Press the **RETURN** or **HOME** buttons to exit the screen.

Important Note: To gang channels quickly press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen and follow the preceding instructions.

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 +20 dBu (G5 audio scale) and +14dBu (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 the levels to the previous setting within half a second. This response is linear.

To adjust this setting in the codec:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press OK.
3. Navigate to **Input IGC** and press OK.
4. Select the channel you want to adjust and press OK.

5. Navigate to the preferred setting and press OK.

9 Configuring AES3 Input Audio

The codec has an **IN1/AES3** input on the rear panel of the codec for AES3 (AES/EBU) format audio. This balanced 110 ohm female XLR input can operate effectively over distances of up to 100 meters and accepts both mono and stereo AES3 signals. If your codec supports multiple channels of AES audio use the DB25 **EXPANSION AUDIO I/O** connection to send/receive up to 4 additional channels of AES3 audio. Please note that expansion inputs/outputs 3, 4, 5 and 6 must be either all analog or all digital AES. See Appendix B for the DB25 wiring pin-outs.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Select **Input Type** and press the button.
4. Navigate to the inputs you want to configure and press the button to toggle between **Analog** and **AES3**.

The 3 pin male XLR **AES3 OUT** connector is capable of sending both mono and stereo AES3 signals.

Important Notes:
- There is a maximum of 6dB of additional gain available when adjusting a AES3 digital input.
- If you switch back to the analog input setting after selecting AES3, the previous analog settings will be recovered.

AES3 Sample Rate Conversion

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 channel 1 and 2 AES3 input sources to 96kHz sampling unless your audio source uses a 44.1kHz sample rate.

Important Notes: If your codec supports up to six AES3 inputs, the three AES3 stereo inputs are multiplexed onto analog inputs 1, 3 and 4. There are also three AES3 stereo outputs via the **AES3 Out** XLR and two are multiplexed onto outputs 3 and 4 (see Appendix B for pin-outs of inputs 3 and 4 and outputs 3 and 4).

The inputs have asynchronous sample rate converters to facilitate sources with different sample rates being attached to the codec. For example, **IN1-AES** (Inputs 1 and 2) can be 48kHz AES-IN2 (IN3&4) can be 32kHz and AES-IN3 (IN5&6) can be 96kHz

Output Sample Rate Converter

The sample rate of the AES3 output is configured using the clock source setting via the **SETTINGS** button and then **Audio > Input Type > AES3 Out**. This configures the sample rate frequency of all AES3 output signals and there are three possible settings.

Locked to AES3 Input
If this setting is used, the codec will use the sync information received by the AES3 input to set the output sample rate in the codec (Note: this is the same as the **AES Rx Clock** setting in Tieline G3 codecs). The codec will initially try to use the signal on AES inputs 1 and 2 as the clock to which the AES outputs are synchronized. If unavailable, it will then attempt to use inputs 3 and 4 or inputs 5 and 6 in that order. If you select this option, all AES inputs must always be synchronized to the same clock source, e.g. if AES3 inputs 1 and 2 use 48kHz sampling then inputs 3 and 4 and inputs 5 and 6 must also be synchronized to the same clock.

Wordclock Sync In
This setting configures the codec for a word clock source via the **SYNC INPUT** on the codec rear panel (this is the same as the **External Word Clock** setting in Tieline G3 codecs). Often this will be a studio reference signal (D.A.R.S., or Digital Audio Reference Signal). In television broadcasting facilities, the audio reference signal should be locked to the video reference if there is one available. The sample rate being received is recognized by the codec and automatically adjusted within it. Sample rates from 32 kHz to 96 kHz are accepted, including the most popular rates of 32 kHz, 44.1 kHz and 48 kHz.

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

10 Merlin PLUS Headphone/Aux Output

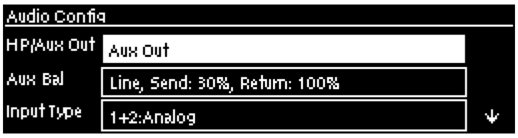
The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for monitoring inputs and return audio. If you are using analog inputs or digital inputs you will see audio metering on the PPMs and can monitor it with the headphones.

> **Important Note:** The front panel **HEADPHONE** output and rear panel **HP/AUX** output share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

Configure for Headphone and Aux Output

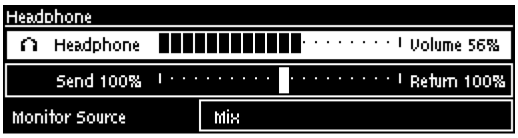
Both the front panel **HEADPHONE** and rear panel **HP/AUX** outputs are configured as stereo headphone outputs by default. To adjust this setting:

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Select **HP/Aux Out** and press to toggle between **Headphone** and **Aux Out**.



Adjust Headphone Output Settings

1. Press the **HEADPHONE** button to display the headphone monitoring adjustment screen.
2. Use the left or right navigation buttons to adjust the volume level up or down. The screen displays level adjustments in real-time.

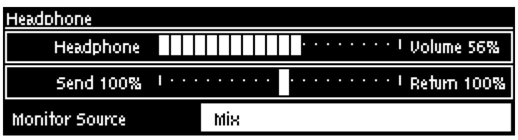


3. Press the down navigation button to select the **Send/Return** audio balance and use the left or right navigation buttons to adjust the balance. The **Send/Return** audio balance dictates whether the front panel **HEADPHONE** output and the rear panel **HP/AUX** output monitors send (input/encoder) audio only, return audio only (decoder audio from a connected device), or a mix of both send and return audio.
4. Press **RETURN** when you have finished to exit the menu.

Note: Headphone levels can also be adjusted by pressing the **SETTINGS** button, navigate to **Audio** and then **HP Vol/Bal** and press .

Adjusting the Monitor Source

In headphone listen mode it is possible to select monitoring sources via **HEADPHONE** **> Monitor Source > [Select audio Source]**.



Navigate to the source you want to monitor and press . Options include:

1. **Mix**: monitors the default program headphone mix, or a customized mix configured using the Matrix Editor. See Matrix Editing for more info.
2. **Audio Stream**: monitors the selected codec audio stream.
3. **Inputs**: monitors the codec inputs (i.e. encoders).

Important Note: PPMs always reflect the default program settings listed in the table below and do not reflect customized mix adjustments using the **Matrix Editor**.

The default headphone mixes for factory programs are displayed in the following table.

Codec Programs	Left	Right
1 x Peer-to-Peer Mono	Inputs 1&2/ Outputs 1&2	Inputs 1&2/ Outputs 1&2
1 x Peer-to-Peer Stereo	Input1 /Output 1	Input 2/Output 2
2 x Mono Peer-to-Peer	Input 1/Output 1	Input 2/Output 2
3 x Mono Peer-to-Peer	Inputs 1 & 3 / Outputs 1 & 3	Inputs 2 & 3 / Outputs 2 & 3
Stereo + 4 x Mono Peer-to-Peer	Input 1 / Output 1	Input 2 / Output 2
6 x Mono Peer-to-Peer	Input1 /Output 1	Input 2/Output 2
1 x Mono Peer-to-Peer + IFB	Inputs 1&2/ Outputs 1&2	Aux In/Aux Out/ Input3/Output3
1 x Stereo Peer-to-Peer + IFB	Inputs 1&2/ Outputs 1&2	Aux In/Aux Out/ Input3/Output3
2 x Mono Peer-to-Peer + IFB	Inputs 1&2/ Outputs 1&2	Input 5/Output 5
2 x Stereo Peer-to-Peer + IFB	Inputs 1&2/ Outputs 1&2	Input 5/Output 5

Adjust Auxiliary Output Settings

Settings for the auxiliary output audio are adjusted similarly to the **HEADPHONE** output, except that the output level is fixed at line level. Configure the front panel **HEADPHONE** output and rear panel **HP/AUX** output as an **Aux Out** and then:

1. Press the **HEADPHONE** button to display the aux output adjustment screen.
2. Use the left or right navigation buttons to adjust the **Send/Return** audio balance.

3. Press **RETURN** when you have finished to exit the menu.

Note: Send/Return balance can also be adjusted by pressing the **SETTINGS** button, navigate to **Audio** and then **Aux Bal** and press .

11 Inserting Hardware Modules

Two slots are available for inserting optional ISDN or POTS connection modules into the codec. The module slots are numbered as follows.

Inserting or Removing a Module

Ensure the codec is not powered up when inserting or removing modules. Where possible use anti-static precautions to help minimize the chance of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

1. Remove power from the codec and then remove the 4 screws from the blanking panel or module installed in the codec.
2. Carefully slide the new module into the module slot and ensure the base of the module remains flat during insertion, to ensure it lines up correctly with the module connector within the codec.
3. Reinsert the 4 screws to hold the module firmly in place.
4. Power up the codec.
5. Press the **SETTINGS** button to verify it is installed correctly.
6. Navigate to **Modules** and press the button.
7. The newly installed module should be visible as **Module 1** or **Module 2**.

Important Note: If the module does not appear in the **Modules** menu in the codec, it is possible that the connector on the module has not lined up correctly with the connector inside the codec. Remove the module and reinsert it carefully to resolve this issue.

12 About ISDN Modules

ISDN stands for Integrated Services Digital Network. The Basic Rate Interface (BRI) of ISDN consists of 2 bearer (B) channels at 64 kbps each and 1 data (D) channel at 16 kbps, i.e. (2B +D). This can be provided over a 2 wire facility and the two B channels can be bonded together to form a single 128kbps channel. The B channel can carry user information such as voice, video or data. The D channel carries signaling information between a user and the network.

Tieline codecs fitted with an ISDN G5 module can provide high quality mono or stereo audio over a single B channel using the Tieline Music algorithm. If you have 2 B channels you can use one as a standby, or configure higher bandwidth mono or stereo connections using algorithms such as MusicPLUS and MPEG. The codec has two module slots available. Each module supports 2 B channels and it is possible to insert two ISDN modules and bond 4 B channels together. This will increase connection bandwidth to 256 kbps for connections using high quality algorithms like aptX Enhanced.

Important Considerations

There are several considerations when using your codec in ISDN mode, including:

- Will you operate within North America or other countries?
- Will you use a single B channel, 2 B channels, or 4 B channels?
- Which network are you using?
- Is your ISDN line Point-to-Point or Point-to-Multipoint?
- What are your directory numbers (DN)?
- If you are in the US, what are your Service Profile ID (SPID) numbers?
- What is your Multiple Subscriber Number (MSN) if you need to enter this outside North America?

ISDN configuration is influenced by the country in which you operate. For example, a SPID does not need to be entered into a Tieline codec for operation within Europe, but it does in North America.

U and S/T ISDN Interfaces

In North America the telephone company provides its BRI customers with a U interface. The U interface is a two-wire (single pair) interface from the phone switch. It supports full-duplex data transfer over a single pair of wires, therefore only a single device can be connected to a U interface.

The situation is different in Europe, the UK, most of Asia, Australia, Africa and parts of the Middle East, where the phone company is allowed to supply the NT-1 and the customer is given an S/T interface. The NT-1 is a relatively simple device that converts a 2-wire U interface into the 4-wire S/T interface.

If you have an NT-1 device connected to the U interface line then you will require a Tieline Euro ISDN G5 module (S/T interface - model: TLISDNEUROG5). If you don’t have an NT-1 device installed then the Tieline US ISDN G5 module (U interface - model: TLISDNUSG5) will be required. You can ring your telecommunications provider to ask if you’re not sure. Note: In Japan use the Tieline Euro ISDN module.

Important Note: Tieline S/T Euro ISDN G5 modules do not have internal terminating resistors. When you connect terminating equipment such as a Tieline codec to an NT-1, 100 ohm termination resistors must be connected between pins 3 and 6 and between pins 4 and 5 at the last socket on the ISDN line. Check your NT-1 device user manual as this may be supported. Suppliers of electronic components sell suitable plugs with termination resistors when required. Please note: U interface ISDN terminations do not require terminating resistors.

How to Configure ISDN G5 Modules

To configure the codec to dial using ISDN for the first time:

1. Ensure that the correct country setting is configured in your codec via **Settings > System > Country**.
2. If you are dialing between two Tieline codecs you normally only need to configure an ISDN dialing program via **Connect > ISDN**. Click here to adjust settings using the HTML5 Toolbox Web-GUI.

Other more advanced settings can also be configured:

1. Select **Settings > Modules > [Select ISDN Module]** to adjust ISDN module settings specific to your codec site. See Configure ISDN Module Settings for more information. Click here to adjust settings using the HTML5 Toolbox Web-GUI.
2. ISDN answering can be configured to suit:

   - Hardware available in the codec, i.e. the number of B channels available.
   - Expected dialing behaviors, e.g. if B channels should bond or not, and whether audio streams need to use **Route** tags.
   - The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline (sessionless calls).

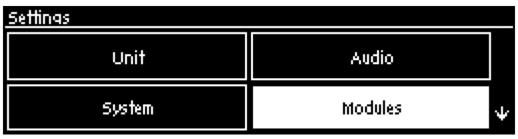
   Adjust answering configuration via **Settings > Answering > ISDN Answer Configs > [Select Config]** and see ISDN Answering Configuration for more information, or click here to adjust settings using the HTML5 Toolbox Web-GUI).

12.1 ISDN Module Settings

ISDN settings in the **Module** menu will determine how each installed module operates at a particular site. Other answering-related settings are available in the **Answering** menu via **Settings > Answering > [Select ISDN Config]**.

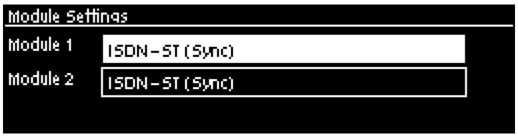
Configuring ISDN G5 Modules

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



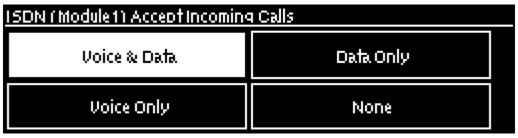
Important Note: You can also configure your ISDN module by pressing the **HOME** button to return to the **Home** screen and select **Connect > ISDN**. Then use the down navigation button to select **Module Configuration** and press the button.

2. Navigate to the module you want to configure and press the button. Note: Module 1 is on the left when looking at the codec rear panel.



Important Note: ISDN Sync should be displayed when an ISDN line is connected to the codec. This appears regardless of whether you have configured the 'ISDN Line Type' correctly.

3. Navigate to **Accept** and press the button. This menu is a call filter to allow or deny voice or data calls according to your preferences. The default setting allows both **Voice & Data**. Select your preferred option and press the button.

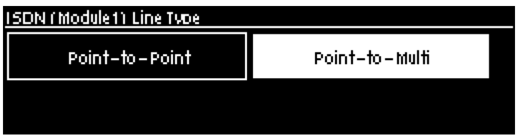


Important Note: G.711 is the default algorithm for incoming connections when **Voice Only** is selected. There are two G.711 algorithms and the one used by the codec depends on the country setting in the codec. The µ-law algorithm is used in the USA, Japan and Canada, whereas the A-law algorithm is used in other countries.

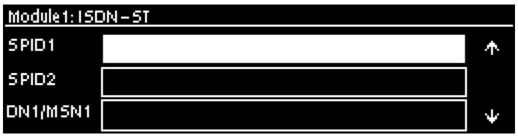
4. Navigate to **Network** and press the button. Select the **Network Type** corresponding to the region in which you are using the codec, then press the button.

Networks	Select
US-Nat	If switch type is National ISDN-1 and 2
US-AT&T	If switch Type is AT&T 5ESS
EU-ETSI	If Switch Type is ETSI (UK, Europe, Australia and most other countries)
JPN-NTT	If you are in the Japan and your network is NTT

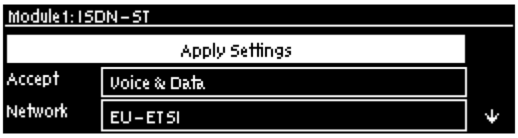
5. Navigate to **Line Type** and press the button. Ask your Telco whether your ISDN line is Point-to-Point or Point-to-Multipoint. By default select **Point-to-Multipoint**, unless your switch type is point-to-point, your Telco says the line is point-to-point, or you are connected to a PABX system. Most PABX systems are point-to-point. Next, press the button.



6. If you are in the US enter DN and SPID numbers as required, or in other regions enter DN or MSN numbers as required. Navigate to each **DN**, **SPID** or **MSN** and press the button before entering each number, then press the button to store each number.



8. Navigate up to **Apply Settings** and press the button to apply all module settings.



Important Notes:

Directory Numbers and Multiple Subscriber Numbers
Directory Numbers (DN) in North America and Multiple Subscriber Numbers (MSN) in the rest of the world are simply phone numbers associated with an ISDN B channel, like lines listed in a typical phone directory. Your Telco will normally supply 2 DN/MSN numbers for each pair of B channels. However, these numbers may or may not be associated with a specific B channel.

Often broadcasters prefer to predict which B channel will answer an incoming call to ensure audio routing is consistent. However, if a DN or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number. Programming DN/MSN numbers for each B channel allows the codec to ignore calls without matching DN/MSN numbers. This is the best way to answer calls from codecs in a predictable manner.

SPID Numbers in North America
ISDN relies on an initialization procedure for associating Service Profiles with specific terminating equipment (e.g. your audio codec) rather than lines. In the US Telcos assign a Service Profile ID (SPID) number which assists in identifying different ISDN services across the network. Your Telco must provide a SPID for each B channel you order when connecting over US-Nat or US-AT&T networks in the US. A SPID is not required when using the AT&T PTP protocol.

Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch. The switch then associates the SPID with a specific Service Profile and directory number.

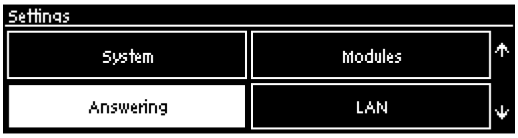
Note: SPID numbers normally include the phone number and additional prefix or suffix digits up to 20 digits long.

12.2 ISDN Answering Configuration

Important Note: For more detailed information about **ISDN Answer Config** parameters, including bonding and 'route' configuration etc., please see Configuring ISDN Answering in the HTML5 Toolbox Web-GUI manual.

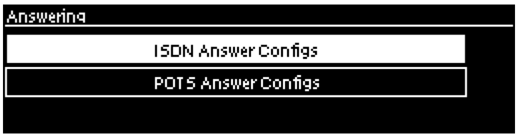
ISDN Answering Configs are used to determine how codec ISDN modules will behave when answering ISDN calls.

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

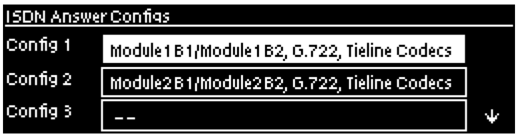


Important Note: You can also configure your ISDN module by pressing the **HOME** button to return to the **Home** screen and select **Connect > ISDN**. Then use the down navigation button to select **Module Configuration** and press the button.

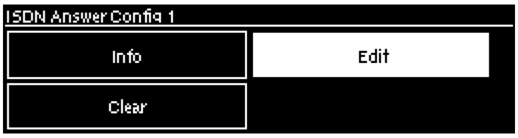
2. Navigate to **ISDN Answer Configs** and press the button.



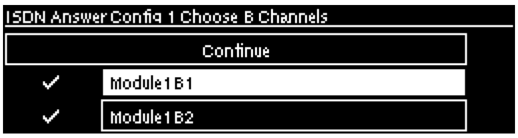
3. Navigate to one of the four available **Configs** and press the button.



4. Navigate to **Edit** and press the button.

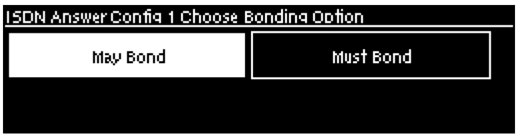


5. Navigate to each B channel and press the button if you want to select/deselect a B channel within the selected **Config**. Navigate to **Continue** and press the button. Notes: The tick symbol confirms a B channel has been selected.



Important Note: If a B channel has been selected within another **Config** it will not be visible. Only available B channels are displayed.

6. Select the bonding method if multiple B channels have been selected, then press the button.



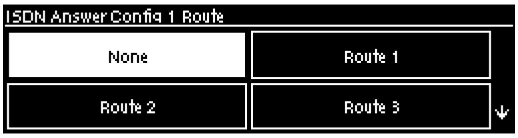
7. Select **Disable** when connecting to Tieline codecs using session data, or select **Enable** if connecting to non-Tieline codecs only, then press the button.

Important Note: Select **Disable** if the codec is expected to receive ISDN calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, once the codec answers a call, it expects to receive Tieline session data from the caller and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (i.e. a non-Tieline codec is calling, or a Tieline codec with session data disabled), it will use the settings in the **ISDN Answering Config** instead.

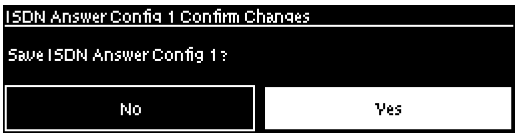
Enable **Sessionless Only** when answering ISDN calls from non-Tieline codecs only. When **Sessionless** is configured, the codec will not wait for the session data. This reduces the time taken to answer an inbound sessionless call.

8. Select the default algorithm when receiving a call from a non-Tieline codec, then press the button.

9. Specify the audio stream **Route** when receiving a call on the answering codec from a non-Tieline codec, then press the button.



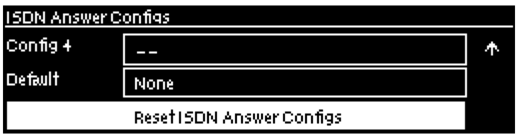
10. Select **Yes** and then press the button to confirm all changes.



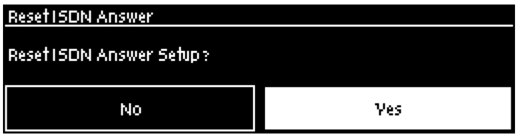
Reset ISDN Answer Configs

To reset ISDN answering settings to factory defaults:

1. Navigate to **Settings > Answering > ISDN Answer Configs > Reset ISDN Answer Configs** and then press the button.



2. Select **Yes** and press the button.


Reset ISDN Answer
Reset ISDN Answer Setup?
No
Yes

13 About POTS Modules

The Tieline POTS G5 module can be used in the codec to stream high quality audio over a POTS (PSTN) phone line. The codec has two module slots available and it is possible to insert one or two POTS G5 modules, or a mix of POTS G5 and ISDN G5 modules as required. The Tieline Music algorithm can deliver 15 kHz quality bi-directional audio at bit rates as low as 24kbps over a POTS connection.

Modem Negotiation and Line Quality

The codec can send and receive high-speed digital information over a standard POTS telephone line via the modem in the POTS G5 module.

G5 POTS modems initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link. The modem then perform a process called ‘training’, during which the codecs at each end of the link analyze the line. The codecs will then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%.

The POTS G5 module contains a SmartDAA™ (Smart Data Access Arrangement) line interface, which isolates the modem from voltages on phone lines. It is important to select the correct country in the codec from which you are dialing. This allows the SmartDAA to automatically adjust for the line voltage present in that country.

Important Note: It is possible to connect two concurrent POTS connections, however the codec will not bond two POTS connections.

Connecting to G3 Codecs using POTS

The codec will successfully connect to Tieline Commander G3 and i-Mix G3 codecs over POTS. These Tieline G3 codecs may use:

- **POTS** modules (older superseded version)
- **POTS G3** modules (current version)

Connecting to POTS G3 Modules

POTS G3 modules operate in the same way as POTS G5 modules when connecting, e.g. they establish a link at the default bit rate of 28.8kbps and then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%.

Connecting to Legacy POTS Modules in G3 Codecs

These modules have slightly different characteristics when connecting. When dialing from a POTS G5 module to these older POTS modules the codecs will attempt to connect initially at 19.2kbps. If line quality is above 80% at this bit rate then the codec will ‘retrain’ the connection up to a maximum of 28,800bps (depending on modem handshaking). The codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

How to Configure POTS G5 Codec Connections

To configure the codec to dial using POTS for the first time:

1. Ensure that the correct country setting is configured in your codec via **Settings > System > Country**. This ensures the correct settings are used by the codec when making POTS connections.
2. When dialing between two Tieline codecs you normally only need to configure a POTS dialing program via **Connect > POTS**.

Other more advanced settings can also be configured:

1. Select **Settings > Modules > [Select POTS Module]** to adjust POTS module settings specific to your codec site. See Configure POTS Module Settings for more information. Click here to adjust settings using the HTML5 Toolbox Web-GUI.
2. If you are connecting to non-Tieline codecs you may need to create an answering "Config" via **Settings > Answering > POTS Answer Configs > [Select POTS Config]**, which will determine the module used and relevant settings for answering a non-Tieline POTS call. See Configure POTS Module Answering for more information. Click here to adjust settings using the HTML5 Toolbox Web-GUI.

Important Note: The codec has a single analog phone input shared by both modules (default setting is **Off**). This phone input is used to monitor modem tones in **POTS Codec** mode and for receiving audio in **Analog Phone** mode. **Phone Input** settings can be adjusted via **Settings > Audio > Phone Input**.

Making Analog Phone (Voice) Calls

All POTS G5 modules are capable of making analog voice calls. It may be necessary to make an analog call to dial a telephone hybrid, or to use for communications, or because there is no Tieline codec at the other end of the link. Remember analog voice calls are only 3 kHz audio quality. To select analog phone answering mode in a POTS G5 module navigate to **Settings > Modules > POTS > Answer Mode [Analog phone]**.

Important Analog Phone Note: The codec has a single analog phone input which is shared by two POTS G5 modules when installed. As a result, two concurrent analog phone connections are not recommended because both connections share the same input and audio will be accepted from the oldest active connection only.

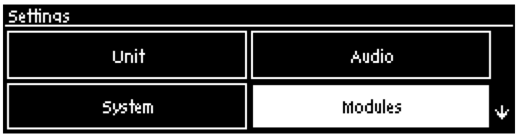
13.1 POTS Module Settings

POTS settings in the **Module** menu determine how your codec behaves at a particular site. Other answering-related settings are available in the **Answering** menu via **Settings > Answering > [Select POTS Config]** if you are connecting to non-Tieline codecs over POTS.

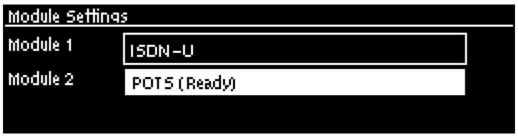
Important Notes: When **POTS (Ready)** is displayed throughout POTS menus it means the POTS module has initialized and is ready to accept or make a call.

How to Configure POTS G5 Modules

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



2. Navigate to the module you want to configure and press the OK button. Note: **Module 1** is on the left when viewing the codec rear panel.



3. Complete configuration changes as per the following options and then navigate up to **Apply Settings** and press the OK button to apply all module settings.

Module (Site) Settings

Answer Mode (Affects Answering Only)

Answer Mode determines how the selected module in the codec will be able to answer incoming POTS line calls. Options include:

- **POTS Codec:** allows the POTS G5 module to receive encoded audio data over a POTS line.
- **Analog Phone:** configures the POTS G5 module to receive a standard analog phone call.
- **Disabled:** disables the POTS G5 module from receiving a **POTS Codec** or **Analog Phone** call.

Calls are answered based on the settings in **Config 1 & 2** via **Settings > Answering > POTS Answer Configs**. Adjustments to these **Config** settings are not normally necessary when connecting between Tieline codecs. Default settings may need to be adjusted when connecting to non-Tieline codecs over POTS (see POTS Answering Configuration for more info).

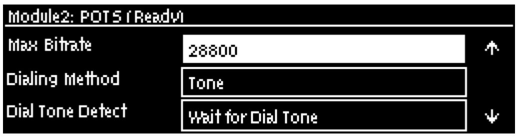
Maximum Bit rate (Affects Dialing and Answering)

The default setting for the **Max Bitrate** is **28800** (28.8kbps) and this only affects **POTS Codec** calls. The range of the setting is 9.6kbps to 33.6kbps. Even if the line is capable of establishing a connection at a higher bit rate, the **Max Bitrate** setting is the highest bit rate that will be attempted.

G5 POTS modems initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

In the initial connection phase, the modems perform a process called ‘training’, to analyze the line and compensate for frequency and phase response. This also cancels out any echo that may be present. The codec will then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%. Negotiation is the process of bit rate adjustment.

Reducing this value can improve connection reliability on poor quality lines. If two codecs are not configured the same, they will always attempt to connect at the lowest of the two **Max Bit rate** settings.



Dialing Method (Affects Dialing only)

Use this menu to select **Tone** (DTMF) or **Pulse** dialing over **POTS Codec** connections. Tone dialing is used always when the **Answer Mode** is **Analog Phone**.

Dial Tone Detect (Affects Dialing only)

There are two settings in this menu:

- **Dial Tone Detect**: The module will only be allowed to dial when a dial tone is present on the line.
- **Blind Dialing**: Allows the module to dial when no dial tone is present.

Monitor Modem Tone (Affects Dialing and Answering)

This setting can be **Enabled** or **Disabled**. If enabled the module will allow audio monitoring of modem tones during connection in **POTS Codec** mode via the phone input. By default, the following phone input monitoring rules apply when multiple POTS G5 modules are installed in a codec and multiple POTS connections are dialed.

Module 1	Module 2	Audio Rule
POTS Codec (Monitor Modem Tone)	POTS Codec (Monitor Modem Tone)	The phone input receives a mix of modem tone audio from both modules
POTS Codec (Monitor Modem Tone)	Analog Phone	The phone input receives analog phone input audio only and mutes modem tone monitoring
Analog Phone	Analog Phone	The phone input accepts audio from the oldest active connection only

Important Notes:

- POTS modem tones are audible in the left side of the headphone output, irrespective of the type of program loaded in the codec.
- Modem tone monitoring will work even if **Phone Input Enable** is **Off** via **Settings > Audio > Phone Input > Phone Input Enable [Off]**.
- Modem tone monitoring is only enabled during the initial connection training and negotiation period in **POTS Codec** mode.
- The monitoring volume can be adjusted via **Settings > Audio > Phone Input > Level**.

Country

This displays the current country setting in the codec. To adjust this setting select **Settings > System > Country**.

Caution: Don't forget to navigate up to **Apply Settings** and press the button to apply all module settings before leaving this menu!

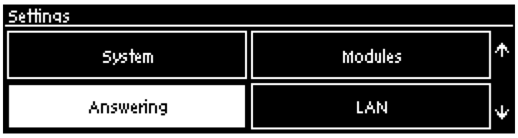
13.2 POTS Answering Configuration

Connection setting preferences are normally exchanged via session data sent between two Tieline codecs when a connection is established. If you answer a call from a non-Tieline codec you will need to create an answering "Config" to determine the settings used when connecting, and designate which module will answer the call (if more than one POTS module is installed).

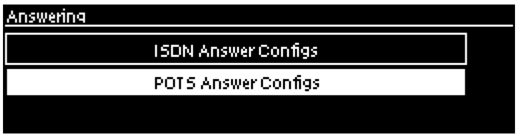
Important Notes:

- **POTS Answer Config** settings are applied to **POTS Codec** connections and not **Analog Phone** connections.
- When receiving a call from a Tieline codec sending session data (i.e. not a **Sessionless** connection), the algorithm setting from the dialing codec overrides the setting in the **POTS Answer Config** menu.
- The default **POTS Answer Configs** accept a call from an incoming Tieline codec with session data enabled. They will also answer a call from a Comrex POTS codec by using the **Other** algorithm.
- For more information about POTS answering parameters, including 'route' configuration, please see Configuring POTS Answering in the HTML5 Toolbox Web-GUI manual.

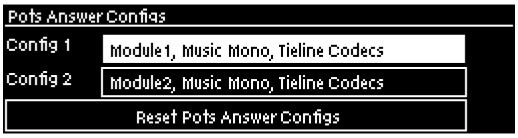
1. Press the the **SETTINGS** button, then navigate to **Answering** and press the button.



2. Navigate to **POTS Answer Configs** and press the button.



3. Navigate to **Config 1** and press to configure a POTS module in module slot 1, or navigate to **Config 2** and press to configure a POTS module in module slot 2.



4. Select **Info** to view current settings or **Edit** to adjust **Config** settings, then press the button.

5. Select **Disable** when connecting to Tieline codecs using session data, or select **Enable** if connecting to non-Tieline codecs only, then press the button.

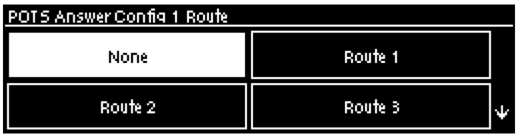
Important Note: Select **Disable** if the codec is expected to receive POTS calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, once the codec answers a call, it expects to receive Tieline session data from the caller and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (e.g. a Comrex POTS codec is calling), it will use the settings in the **POTS Answering Config** instead.

Enable **Sessionless Only** when answering POTS calls from non-Tieline codecs only. When **Sessionless** is configured, the codec will not wait for the session data. This reduces the time taken to answer an inbound sessionless call.

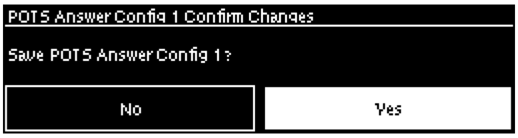
6. Select **Other** when connecting to Comrex® Vector, Matrix® and BlueBox® codecs, then press the OK button.

Important Note: On the Comrex codec select its "Music" algorithm. Please note that 9.6kbps connections are not supported by the Comrex codecs.

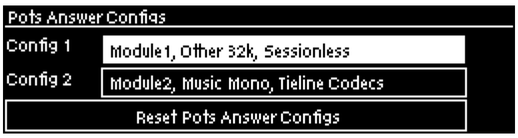
7. If required you can specify the audio stream **Route** when answering a call from a non-Tieline codec, then press the OK button.



8. Select **Yes** and press the OK button to confirm the new **Config** settings.



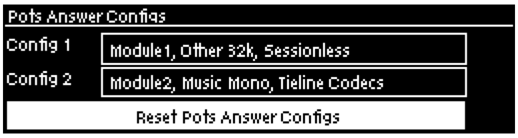
9. The new **Config** will be displayed showing the updated settings.



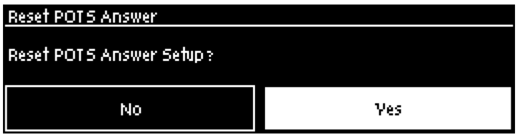
Reset POTS Answer Configs

To reset POTS answering settings to factory defaults:

1. Navigate to **Settings > Answering > POTS Answer Configs > Reset POTS Answer Configs** and then press the button.



2. Select **Yes** and press the button.



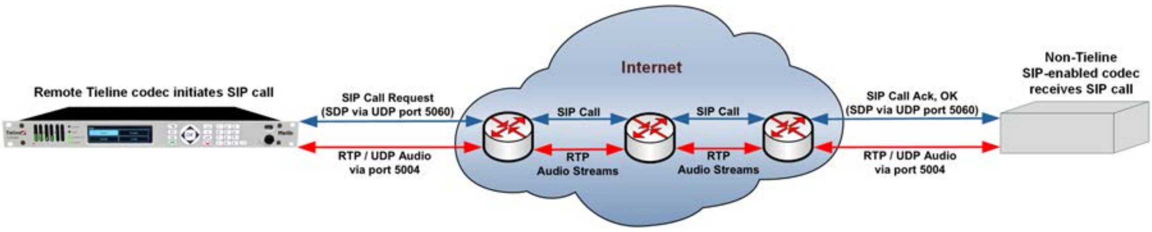
14 About SIP

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.



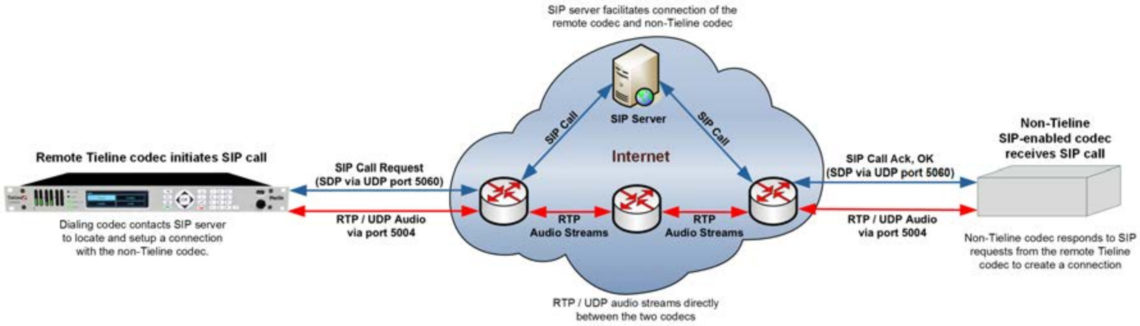
There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP and SDP is used for. The second stage is when data transfer occurs and this is left to the other protocols such as RTP/UDP to stream audio data. SDP works with a number of 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 ports 5004 to 5054.

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 Peer-to-Peer. 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: 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. **ETH1** or **ETH2**.

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 select **SETTINGS** **> 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]**

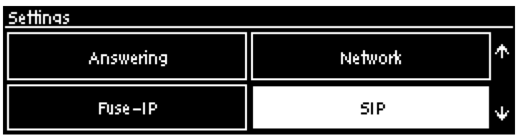
14.1 Configuring SIP Interfaces

The codec supports dialing over two SIP interfaces simultaneously.

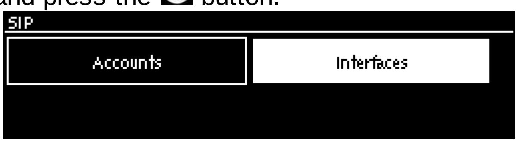
Important Notes:
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **ETH1** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configured to use **ETH2** by default, which is mapped to the **Secondary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **ETH1** or **ETH2**.
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** 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**:

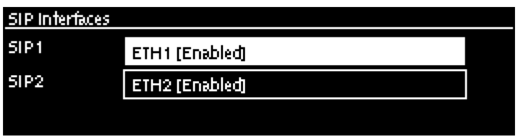
1. Press the **SETTINGS** button, then navigate to **SIP** and press the button.



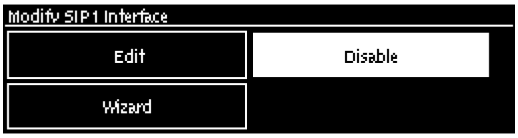
2. Navigate to **Interfaces** and press the button.



3. Select either **SIP1** or **SIP2** and press .

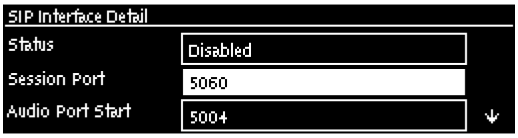


4. Select **Disable** to disable the interface before editing settings.

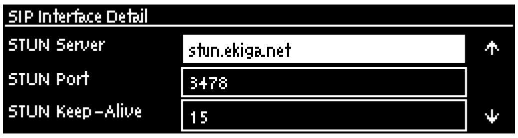


5. Select **Edit** and then press the OK button to individually select and configure SIP interface settings.

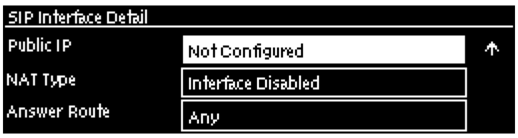
6. Navigate to each field in turn and press the OK button to edit SIP interface settings as required.



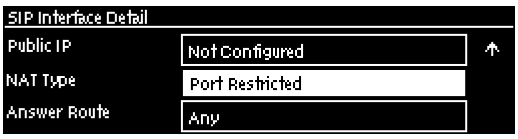
7. Select **STUN Server** and press the OK button to enter server details if using STUN (Session Traversal of User Datagram Protocol). Note: UDP Port 3478 is the default port assigned.



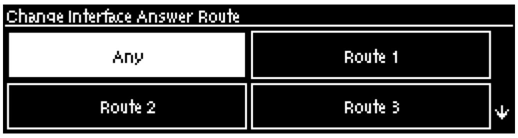
8. Select **Public IP** and press the OK button if you want to dial over SIP from behind a firewall. Enter the public IP address and then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. 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.



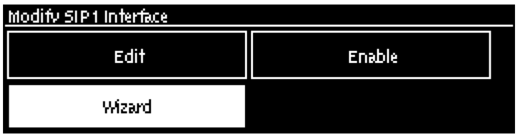
9. The **NAT Type** used by the firewall is updated if STUN has been enabled and the **NAT Type** can be determined.



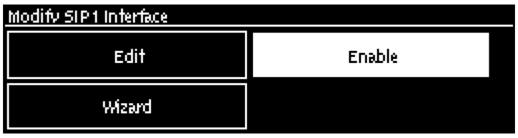
10. Select **Answer Route** and use the **Change Interface Answer Route** screen to route calls using this SIP interface to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded program. If the default value **Any** 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. The SIP interface **Wizard** can also be used to enter SIP account information. Select **Wizard** to navigate through each screen in turn automatically.



12. After completing configuration, navigate to **Enable** in the **Modify SIP Interface** menu and press the button to enable the interface so that it can be used to make a SIP call.



14.2 Configuring SIP Accounts

Getting Started

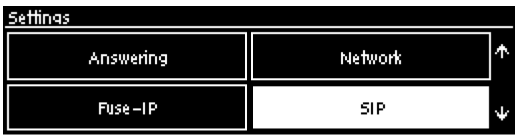
Up to 6 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, select the SIP server to which you will register your codec. On a LAN this may be your own server, or it could be one of the many internet servers available. When you register an account 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** (**ETH1**) or **SIP 2** (**ETH2**).
- 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.



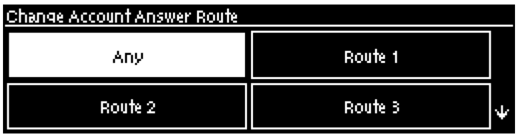
2. Navigate to **Accounts** and press the button.

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

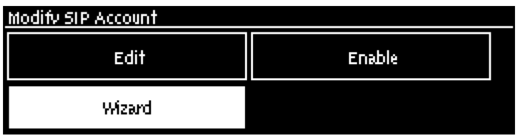
4. Select **Edit** and then press to individually select and enter SIP server account details.

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

6. Select **Answer 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. The SIP account **Wizard** can also be used to enter SIP account information. Select **Wizard** to navigate through each screen in turn automatically.



8. After completing account configuration, select **Enable** in the **Modify SIP Account** menu and press the button to enable the SIP account and register it with the SIP server.

9. To confirm the account has been registered successfully, verify the **USER LED** on the front panel of the codec is illuminated solid orange. Alternatively, 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. Configure this menu setting via **Connect > IP > SIP > Setup > Account.**

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.

The **USER LED** on the front panel of the codec also displays the following SIP account registration states:

USER (SIP) LED State	Description
Not Illuminated	Indicates the account is not yet registered.
Orange Flashes	Codec is in the process of registering to an active SIP server account.
Solid Orange Illumination	The codec has been successfully registered to an active SIP account.

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 **Via** selected in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **ETH1**, **ETH2** or **Wi-Fi**.

15 Language Selection

English is the default language in the codec. To select a new language:

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 .

16 About Program Dialing

What Defines a Program?

Tieline Genie and Merlin codecs use programs to connect to another codec. 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.

Tieline Genie and Merlin codecs operate similarly to Tieline G3 codecs. 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 information about how the codec receiving the call should be configured.
2. Once the codec receiving session data from the dialing codec has received information successfully, it sends an acknowledgment to the dialing codec and streaming can commence.

For example, if you configure a standard stereo program on the dialing codec using a particular algorithm and bit rate settings etc., these settings will be configured on the dialing codec when the codec connects. 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.

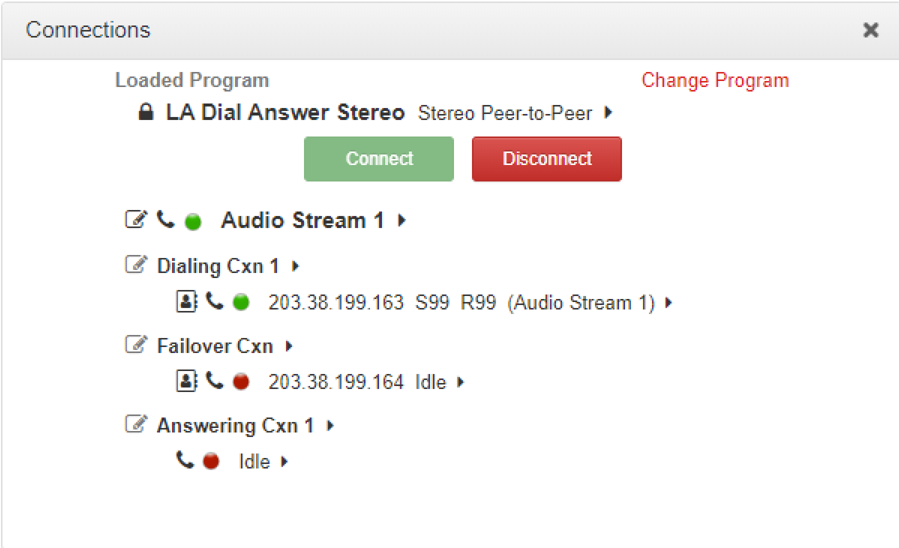
For example, if your routing requirements require the codec at the studio to always connect in mono, simply load and lock a mono program in the codec. Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will be rejected.

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 has its own:

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

The following image displays a simple peer-to-peer program in the **Program Manager panel** within the HTML5 Toolbox Web-GUI, which can be used to configure and edit all program parameters. The following program is configured to send a single stereo audio stream and will allow the codec to both answer and dial if required. A backup dialing connection is configured in case the primary connection fails.



Creating Programs

Only the simplest peer-to-peer (point-to-point) programs can be created using the codec front panel. The HTML5 Toolbox Web-GUI contains a **Program Manager panel** with a wizard for configuring program settings and backup connections. See Using the HTML5 Toolbox Web-GUI for more detailed information on setting up mono, stereo and multiple stream programs. Edit settings easily at the touch of a button and use existing programs as templates for creating other programs.

Mono and Stereo Peer-to-Peer Programs

New peer-to-peer programs can be created using the codec front panel keypad (see Steps to Connect over IP). If you know the IP address of the codec you want to dial then all you need to do is enter this into the codec, select your preferred connection settings and then press **CONNECT** .

Programs dialed from the front panel are automatically saved as **Recent Programs** which retain all audio stream dialing and configuration information. These **Recent Programs** are displayed when you press the **CONNECT** button from within any menu except the **IP Mode** or **SIP Mode** screens, or the **Connect IP** or **Connect SIP** screens.

Ensure you configure all the correct connection settings when using the codec front panel, because these are stored as part of the program's profile when you first connect. They cannot be adjusted afterwards without using the editing features in the **Program Manager panel** within the HTML5 Toolbox Web-GUI.

Important Note: When configuring a connection use the **Save** function to save programs permanently to the codec's **Programs** menu. Otherwise they are stored to the **Recent Programs** list and will be overwritten after several calls have been made.

17 Getting Connected Quickly

Before attempting a new audio stream connection please connect and adjust the following:

1. Attach power to the codec.
2. For IP connections, attach RJ45 Ethernet cables to at least one of the **ETH** ports on the codec's rear panel. Attach cables to ISDN or POTS modules inserted in your codec as required.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the codec's front panel.
4. Check that the correct country is selected in the codec.

i. Press the **SETTINGS** button.
ii. Navigate to **System** and press the button.
iii. Navigate to **Country** and press the button.
iv. Use the navigation buttons to select your country of operation and press the button.

5. Make sure you know the IP address, or line numbers for dialing over ISDN or POTS to the destination codec.

Important Note: It is important to set the correct country setting for connections over POTS to adjust the POTS G5 module for varying ring tones and line impedances in different countries. The country setting also affects whether G.711 µ-Law (North America/Japan) or A-Law (Europe/Australasia) coding is used over IP, SIP and ISDN connections.

17.1 Steps to Connect over IP

The following procedure will create a custom peer-to-peer connection program using the codec front panel keypad and navigation buttons. It instructs how to connect your codec over IP for the very first time without using the Toolbox Web-GUI and your computer for configuration.

Important Notes:

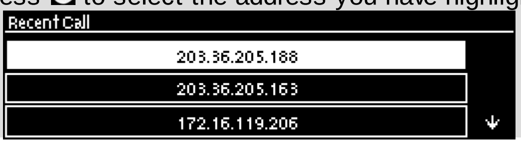
- See Using the HTML5 Toolbox Web-GUI for details on configuring connections remotely via a computer.
- See Installing the Codec at the Studio for valuable information about installing your codec, negotiating firewalls and port forwarding.
- See Tips for Creating Reliable IP Connections for a range of IP information to assist with setting up IP services for your codecs.
- See Testing IP Network Connections to learn how you can test and verify the reliability of your IP connection.

1. Press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
   - Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press 1 on the numeric keypad to toggle channel 1 on and off.
   - Use the up and down navigation buttons to select the gang function and press the OK button to toggle ganging on/off.
   - Use the up and down navigation buttons to select a single channel, or ganged channels. Note: A channel is highlighted when selected.
   - Use the left and right navigation buttons to adjust the input levels up or down.

2. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > Tieline** and press the OK button. Note: Select **SIP** or **Sessionless** instead of **Tieline** if these connections are required.

3. Use the **RETURN** button to delete any numbers if already entered, then 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 (or use the **Virtual Keyboard**). Next, press the down navigation button to select **Setup** and press OK.

Important Note: The codec remembers recent IP addresses just like a cell-phone. To view these addresses just press OK when you select the **Connect IP** screen. The most recent addresses and programs are listed first and you can use the navigation buttons to scroll up and down. Press OK to select the address you have highlighted.



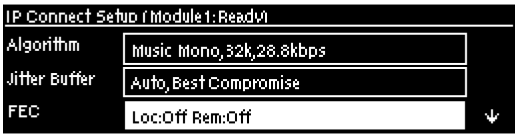
4. Navigate to **Algorithm** and press OK.

5. Use the navigation buttons to select an algorithm profile or manually enter algorithm settings, then press .

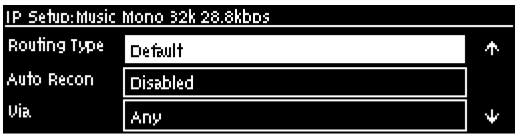
6. If you decide to manually program the algorithm, use the navigation buttons to select your preferred algorithm sample rate (if displayed) and bit rate, pressing after each option is selected.

7. Press the down navigation button to select **Jitter Buffer** and press to select a different automatic jitter buffer setting for your connection, or to enter a fixed buffer setting in milliseconds (maximum 5000 ms). The default **Auto, Best Compromise** setting is a good starting point for most internet connections.

8. Press the down navigation button to select **FEC** (forward error correction) and press to view selection options. Use the navigation buttons to select the FEC percentage you want to use and press .



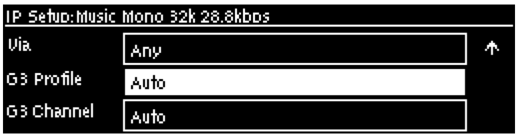
9. Navigate to **Routing Type** to configure routing options as per the table below:



Routing Type Options:	
Default ISDN POTS	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 ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	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.

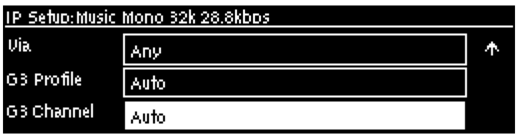
10. Navigate to **G3 Profile** to configure profile settings when dialing a Tieline G3 Codec.



Important Note: The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 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.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

11. 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 this 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.



12. When configuration is complete press the **RETURN** button to navigate backwards to the **Connect IP** screen that the IP address was entered into.

Important Note: At this point you can navigate to **Save** on the **Connect IP** screen and press to use the numeric **KEYPAD** to name the program and press to save the program.

13. Press the **CONNECT** button to make a connection. The **Wait Connecting** screen appears during the connection process.

14. Alternatively, to load a saved program and dial press the **HOME** button, navigate to **Programs**, select the program you want to dial and press the **CONNECT** button to load the program and dial.

15. When dialing, the **CONNECTED LED** on the front of the unit will flash green. When connected, the **CONNECTED LED** on the front of the unit will illuminate solid green.

16. From the **Home** screen use the down navigation button to select **Cxns** and view connection **Status** and press to view connection statistics for IP packets being sent over the connection. To negotiate higher bit rates press F2 then **3** on the numeric **KEYPAD**; for lower bit rates press F2 then **9**.

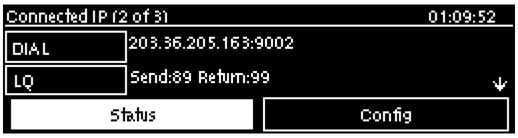
17.2 Monitoring IP Connections

Connection Details

The number of active audio streams and connections is displayed on the **Home** screen via **Cxns**. In the following image two connections (left bracketed number) and two audio streams (right bracketed number) are currently in use.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.
3. If only a single IP audio stream is connected, the **Connected IP** screen displays details of the active connection. When multiple audio streams are connected, navigate to the one you want to view in the **Connections** screen and press the button to view more details.

For IP connections you can view the IP address dialed and the **LQ** (link quality) on the screen. Use the down navigation button to view the algorithm being used, the connection bit rate, total bytes used and the amount of jitter buffer delay over the IP network.



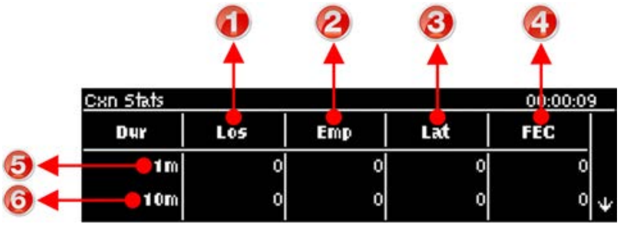
Link Quality (LQ) Readings

Send and return LQ numbers can also help to determine if a problem is occurring at either end of a connection. 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 button to display the **Cxn Stats** (connection statistics) screen. This displays the performance of the codec in 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 used (replaced) when FEC is enabled
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 a packet arrival analysis table with solutions for any noticeable packet loss 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 ISPs 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 dropouts 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 and network is unreliable – renegotiate bit rate and /or FEC down • If many late packets, increase jitter buffer
Late	The number of packets that arrive late and after audio play out.	• Network congestion • Jitter Buffer depth is too low	• Auto-jitter buffer will adjust automatically • 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.

17.3 Steps to Connect over ISDN

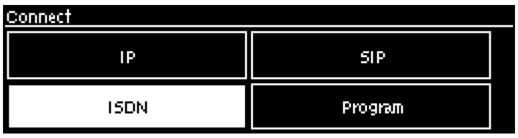
The following procedure explains how to create a custom peer-to-peer program and dial another Tieline codec over ISDN using the front panel keypad and navigation buttons.

Important Notes:
- See Testing ISDN Connections for valuable information about setting up and maintaining reliable ISDN connections.
- See ISDN Module Configuration for details on module settings.
- See ISDN Answering Configuration for details on ISDN answering settings.
- See Configuring ISDN to configure ISDN settings using the HTML5 Toolbox Web-GUI.

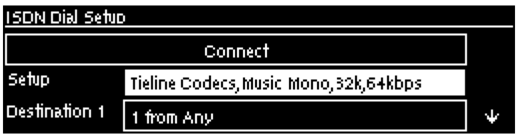
5. For detailed information about connecting with other brands of codec using ISDN visit www.tieline.com.

1. Press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
   - Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press 1 on the numeric keypad to toggle channel 1 on and off.
- Use the up and down navigation buttons to select the gang function and press the button to toggle ganging on/off.
- Use the up and down navigation buttons to select a single channel, or ganged channels. Note: A channel is highlighted when selected.
- Use the left and right navigation buttons to adjust the input levels up or down.

2. Press the **HOME** button to return to the **Home** screen, select **Connect > ISDN** and press the button.



3. Navigate to **Setup** and press the button.



4. Select whether to dial with Tieline Session Data or select **Sessionless** if dialing a non-Tieline codec, then press the button.

Important Note: By default, when Tieline codecs dial they send call configuration settings to the remote codec using Tieline Session Data. This configures the codec receiving the call with matching algorithm, sample rate and bit rate settings. This does not occur when dialing to non-Tieline devices, therefore **Sessionless** must be selected to provide compatibility.

5. Select the **Dial Route** to use for this audio stream if one is required, then press the OK button. Note: See Configuring ISDN Answering for more information on **Dial Route** and **Answer Route** tags. These are useful when routing multiple audio streams over transports like ISDN.

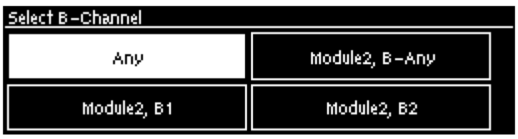
6. Select the number of B channels being used for the audio stream connection, then press the OK button.

7. Select an algorithm, then press the OK button.

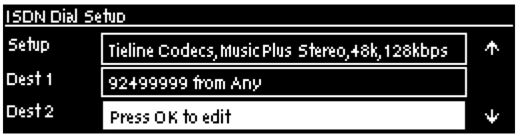
8. Select the sample rate if required, then press the OK button.

9. Select **Destination 1** and press the OK button, then use the numeric **KEYPAD** to enter the ISDN number you want to dial and use the **RETURN** button to delete any numbers already entered. Then press the OK button.

10. Select the preferred B channel or channels to use when dialing and press the OK button.



11. If you are dialing over multiple B channels to create a bonded connection select the next destination, e.g. **Destination 2**, and use the numeric **KEYPAD** to enter the next ISDN number you want to dial. Do this for all B channel destinations.



12. Navigate down to **Auto Reconnect** and press the button to toggle between **Enabled** and **Disabled**. Note: This is normally enabled on the dialing codec only.

13. At this point we recommend you save a program to simplify dialing and to store this configuration for future use. Use the up navigation button to select **Save as Program** and press the button.

14. Use the numeric **KEYPAD** to name the program, then press to save the program.

15. It is possible to dial the B channels associated with this audio stream from this menu. Use the up navigation button to select **Connect** and press to connect.

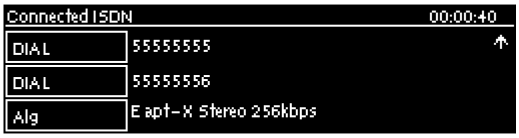
16. When dialing, the **CONNECTED LED** on the front of the unit will flash green. When connected, the **CONNECTED LED** on the front of the unit will illuminate solid green.

Important Note: To load a saved program and dial press the **HOME** button, navigate to **Programs**, select the program you want to dial and press the **CONNECT** button to load the program and dial.

17.4 Monitoring ISDN Connections

Each new audio stream connection becomes visible in the **Cxns** menu via the **Home** screen.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.
3. If a single ISDN audio stream is connected, the **Connected ISDN** screen will display details of the active ISDN connection. When multiple audio streams are connected, navigate to the one you want to view in the **Connections** screen, e.g. ISDN, and press the button to view more details.



17.5 Steps to Connect over POTS

The following procedure explains how to create a custom peer-to-peer program and dial another Tieline codec over POTS using the front panel keypad and navigation buttons.

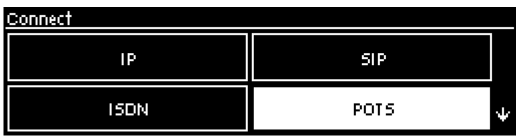
Important Notes:

- See POTS Connection Tips and Precautions for valuable information about setting up and maintaining reliable POTS connections.
- See POTS Module Settings for details on module settings.
- See POTS Answering Configuration for details on POTS answering settings (required for answering calls from non-Tieline POTS codecs)
- See Configuring POTS for details on configuring codec connections via a computer.
- The **Local** and **Remote** line quality displayed for **POTS Codec** connections is related to the actual POTS line quality at either end of the link. This reading affects the maximum allowable bit rate when the codec is training and negotiating a connection.

1. Press F1 and press and release the right arrow button to open the **Input Audio Level** adjustment screen.
   - Press the number on the keypad corresponding to the channel you want to toggle on or off. E.g. press 1 on the numeric keypad to toggle channel 1 on and off.

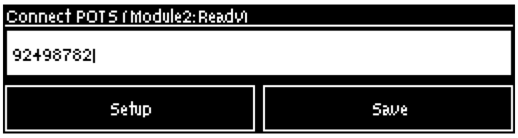
- Use the up and down navigation buttons to select the gang function and press the button to toggle ganging on/off.
- Use the up and down navigation buttons to select a single channel, or ganged channels. Note: A channel is highlighted when selected.
- Use the left and right navigation buttons to adjust the input levels up or down.

2. Press the **HOME** button to return to the **Home** screen, select **Connect > POTS** and press the button.

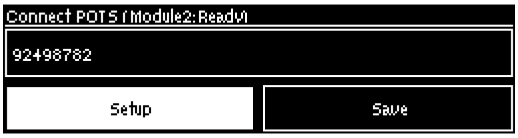


3. Use the **RETURN** button to delete any numbers if already entered, then use the numeric **KEYPAD** to enter the number you want to dial. Note: When **POTS (Ready)** is displayed

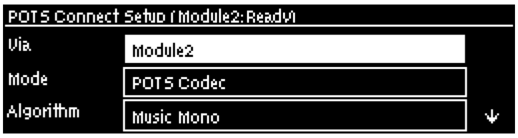
throughout POTS menus it means the POTS module has initialized and is ready to accept or make a call.



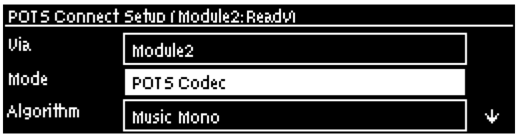
4. Navigate to **Setup** and press the OK button.



5. Select **Via** to nominate the module used when dialing a connection, or select **Any** to use any available module in the codec, then press the OK button.

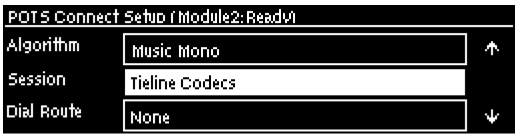


6. Navigate to the connection **Mode** and press the OK button to toggle between selecting either **POTS Codec** or **Analog Phone**. Note: Redundant settings in the menu will disappear If you select **Analog Phone**.



7. Select an algorithm, then press the OK button.

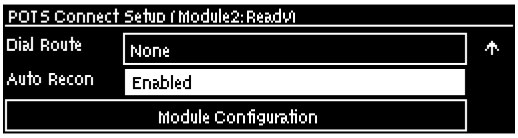
8. Select **Tieline Codecs** session data when connecting to another Tieline codec or **Sessionless** when dialing to non-Tieline POTS codecs.



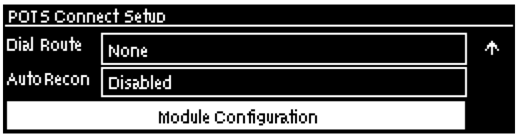
Important Notes: To dial a Comrex® Vector, Matrix® or BlueBox® codec over POTS select the **Other** algorithm and **Sessionless**. Please note that 9.6kbps connections are not supported by Comrex codecs.

9. Select the **Dial Route** to use for this audio stream if one is required, then press the OK button. Note: See Configuring POTS Answering for more information on **Dial Route** and **Answer Route** tags. These can be useful when routing multiple audio streams.

10. Navigate to **Auto Reconnect** and press the button to **Enable** or **Disable** this setting as required.



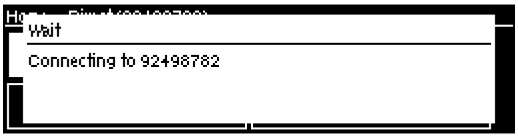
11. Select **Module Configuration** to adjust other settings specific to how you want POTS modules to dial and answer, e.g. Maximum connection bit rate and dial tone settings etc.



12. When configuration is complete press the **RETURN** button to navigate back to the **Connect POTS** screen.

Important Note: At this point you can navigate to **Save** on the **Connect POTS** screen and press to use the numeric **KEYPAD** to name the program. Then press to save the program.

13. Press the **CONNECT** button to make a connection. The **Wait Connecting** screen appears during the connection process.



Note: To load a saved program and dial press the **HOME** button, navigate to **Programs**, select the program you want to dial and press the **CONNECT** button to load the program and dial.

14. To negotiate higher bit rates press F2 then **3** on the numeric **KEYPAD**; for lower bit rates press F2 then **9**.

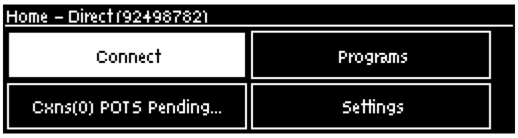
Important Note: To load a saved program and dial press the **HOME** button, navigate to **Programs**, select the program you want to dial and press the **CONNECT** button to load the program and dial.

See Monitoring POTS Connections for more details on monitoring the different POTS connection states.

17.6 Monitoring POTS Connections

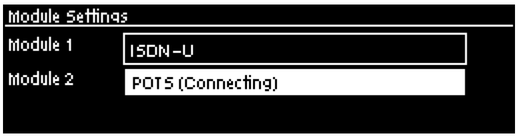
Monitoring POTS Calls when Dialing and Connecting

1. When dialing and connecting:
   - The **CONNECTED LED** on the front of the unit will flash green.
   - The **Cxns** section on the **Home** screen displays **Pending** while the call is connecting (prior to streaming audio data).



Cxns Displays Pending

   - **Connecting** is displayed in the **Modules** menu via **Settings > Modules**.



Modules Displays Connecting

2. While connecting you can also monitor dial tones and modem handshaking etc., via the left channel of the headphone output. For more details see the **Monitor Modem Tones** section in POTS Module Settings.

Monitoring POTS Calls when Connected

1. The **CONNECTED LED** on the front of the codec illuminates solid green when connected.

2. The newly connected audio stream connection becomes visible in the **Cxns** menu via the **Home** screen. To view connection details:
   a. Use the down navigation button to select **Cxns** and press the button.
   b. When multiple audio streams are connected, navigate to the one you want to view and press the button to view connection details.

The **Local** and **Remote** line quality displayed for **POTS Codec** connections is related to the actual POTS line quality at either end of the link. This reading affects the maximum allowable bit rate when the codec is training and negotiating a connection. It also indicates the stability of the connection when a call has been connected for a long period of time. If the line quality starts drop quite low after being connected for a long period, we recommend you retrain the connection to improve the line quality and avoid loss of audio.

17.7 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. Use the navigation buttons to select **Programs** and press the button.
3. Use the up and down navigation buttons to select the program you want to use, then press the **CONNECT** button to load the program and make a connection.
4. The **Wait Connecting** screen appears during the connection process and then connection details are displayed.

17.8 Disconnecting a Connection

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

17.9 Redialing a Connection

Press the **CONNECT** button from any codec menu to redial previous connections (except menus accessed via the **Connect** > **IP**, **ISDN** or **POTS** screens).

Manually dialed connections are saved as programs - retaining all the dialing and configuration information programmed into the codec. A program is identified in the **Recent Programs** redial screen using either a previously entered name, or by a dialing address or number (manually dialed connections).

17.10 Configuring Auto Reconnect

Auto Reconnect is disabled by default. When enabled, the dialing codec attempts to reconnect if data is temporarily lost over a 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 web-GUI.

Auto Reconnect using IP

1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **IP** and press the button.
2. Select the **IP Session** mode you are using to connect.
3. Select **Setup** and press .
4. Navigate to **Auto Recon** and press to toggle between **Enabled** and **Disabled**.

Auto Reconnect using ISDN

1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **ISDN** and press the button.
2. Navigate to **Auto Recon** and press to toggle between **Enabled** and **Disabled**.

Auto Reconnect using POTS

1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **POTS** and press the button.
2. Select **Setup** and press .
3. Navigate to **Auto Recon** and press to toggle between **Enabled** and **Disabled**.

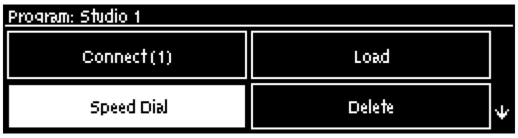
17.11 Speed Dialing Connections

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 you want to assign a speed number to and press the button.
4. Navigate to **Speed Dial** and press the button.

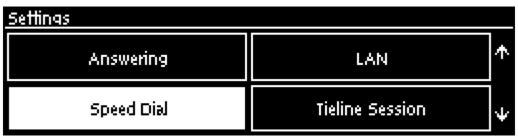


5. Navigate to the program you want to assign a speed dial number, then press the button.
6. A confirmation message will display the number assigned.

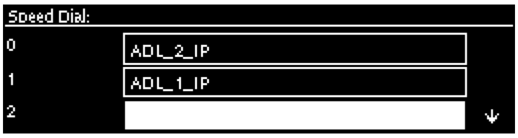
Assigning and Viewing Speed Dials via the Speed Dial Menu

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

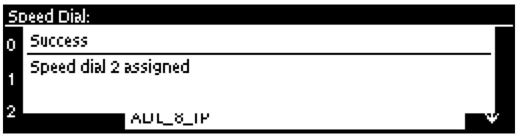
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Speed Dial** and press the button.



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.



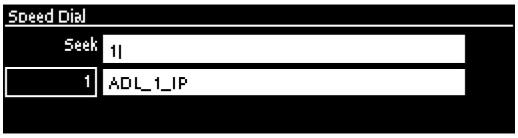
Speed Dialing

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, or speed dial numbers 0-9 if preferred.

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.



17.12 Dial/Disconnect Multiple Audio Stream Programs

Multiple Audio Streams within Programs

Some programs are created to allow simultaneous audio stream connections with different destination codecs, e.g. 2 x Mono peer-to-peer programs. These programs can only be created using the Toolbox web-GUI.

There are two ways to simultaneously dial multiple audio stream connections within these types of programs:

1. Load the program into the codec via the front panel and dial.
2. Connect to the codec using the Toolbox HTML web-GUI. Programs can be loaded from within the **Program Manager panel** or **Connections panel**. Use the **Connections panel** to connect.

Dialing Multiple Audio Stream Programs with the Front Panel

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.
3. Use the up and down navigation buttons to select the program you want to connect with, then press the **CONNECT** button to make a connection.
4. The **Wait Connecting** screen appears briefly and then the **Home** screen is displayed.

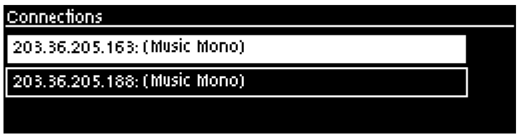
It is also possible to redial the connection, see Redialing a Connection for more information.

Disconnect All Audio Stream 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.

Disconnect a Single Audio Stream

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.
3. Use the up and down navigation buttons to select the connection you want to disconnect.



3. Press the red **DISCONNECT** button on the numeric **KEYPAD**.
4. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the button to confirm the disconnection.

17.13 Dialing SIP Connections

Important Note:
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **ETH1** by default.
3. **SIP2** is configured to use **ETH2** by default.
4. It is not possible to renegotiate the connection bit rate over a SIP connection.
5. When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level. To do this select **SETTINGS** **> Audio > Ref 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]**
5. For detailed information about connecting with other brands of codec using SIP visit www.tieline.com.

Dialing Peer-to-Peer 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. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
2. Use the **RETURN** button to delete any numbers if already entered, then 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 (or use the **Virtual Keyboard**).

3. Press the down navigation button to select **Setup** and press to adjust the algorithm, jitter buffer and encode/decode direction if required.
4. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
5. Press the **CONNECT** button to make a connection.

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 already entered. Alternatively, if you have dialed the SIP address previously, press the **RETURN** button to view the **Recent Call** screen and select the SIP address you want.

Dialing Using a SIP Account

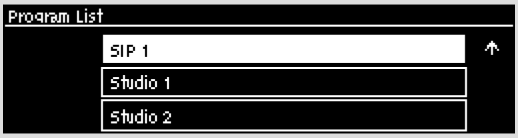
To dial using a SIP account you first need to add a SIP account and register it to the codec. See Configuring SIP Accounts for more info.

1. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > SIP**.
2. Then press the down navigation button to select **Setup** and press .
3. Navigate down to **Account** and and press .

4. Navigate to a registered SIP account and press .

5. 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 entering SIP account details into the codec.
- If you don't save the program during configuration, a temporary program is created after you dial the SIP connection for the first time using the codec **KEYPAD**. The temporary program will appear in the recent calls list if you want to redial the program.



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

17.14 Creating a Multicast Client 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.
- The default UDP audio port is 9000 for a multicast client program configured via the codec front panel.
- 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.

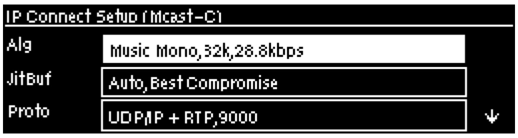
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Backup and Restore Functions.
- To learn more about programs see the section titled About Program Dialing.
- Relay functionality is available for multicast connections between Tieline codecs.
- See Toolbox web-GUI documentation for more detailed information about configuring multicast client programs.

1. Press the **HOME** button to return to the **Home** screen, select **Connect > IP > Sessionless** and press the button.

2. Select **Multicast Client** to configure a client codec program.

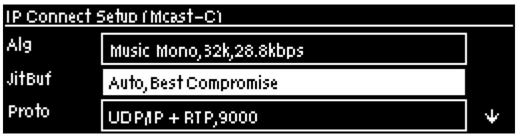
3. Use the **RETURN** button to delete any numbers already entered, then use the numeric **KEYPAD** to enter the multicast IP address you want to dial, using the or buttons to enter the periods in the IP address. The same multicast address and audio port must be used for both the server and client programs. Next, press the down navigation button to select **Setup** and press .

4. Press the down navigation button to select **Algorithm** and press .



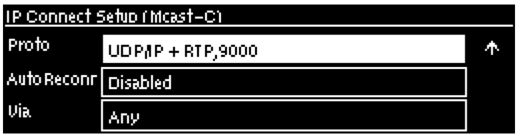
5. Use the navigation buttons to select an algorithm profile, or manually select algorithm settings, then press .

6. Click to configure the Jitter Buffer from 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 .

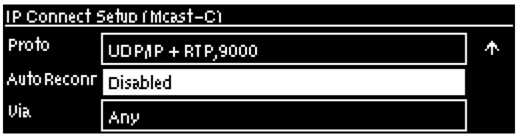


Important Notes: Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

7. Select **Protocol** to select the audio protocol and adjust the **Local Audio Port**. Select **UDP/IP +RTP** for RFC compliant IP streaming. Press OK to save settings.



8. If required, enable **Auto Reconnect** and use **Via** to specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.



9. Press the **RETURN** button when configuration is complete to navigate backwards to the **Connect IP** screen that the multicast IP address was entered into.

Important Note: At this point you can navigate to **Save** and press OK to save the settings as a custom program for subsequent recall and dialing. Use the numeric **KEYPAD** to give the program a name and press OK to save the program. A confirmation message is displayed after the program is saved.

Connecting a Multicast Client Program

1. After you have created multicast server and client programs on your codecs you can dial multicast connections. First select the multicast server program you want to use on the server codec and dial to connect.

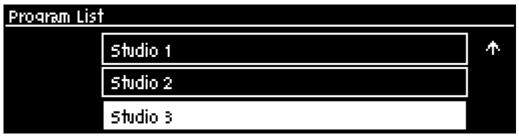
2. Select and load the multicast client program on each of the multicast client codecs and dial the multicast IP address to begin receiving multicast audio packets.

   a. Press the **HOME** button to return to the **Home** screen.
   b. Use the navigation buttons to select **Programs** and press the OK button.
   c. Use the up and down navigation buttons to select the multicast client program you want to connect with, then press the OK button to load the program.
   d. Press the **CONNECT** button to make a connection.

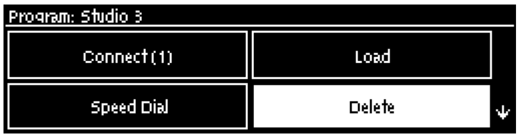
You can navigate to **Cxns** on the **Home** screen to view a codec's connection **Status**, then press to view connection statistics for IP packets being received over the connection.

17.15 Deleting Programs

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 you want to delete and press the button.



4. Navigate to **Delete** and press the button.

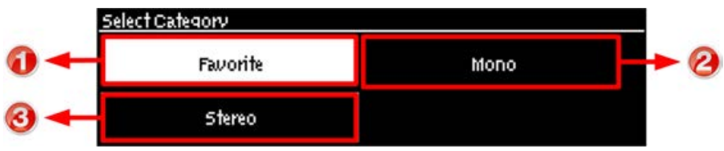


5. Confirm the deletion and press the button.

17.16 Selecting Algorithm Profiles

A number of pre-programmed mono and stereo dialing profiles are available for programming the codec quickly without individually selecting algorithms and bit rates etc. These profiles have been programmed with the most popular settings that provide high quality connections using each available algorithm.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** session mode and press the button. Note: algorithm profiles are only available for Tieline session connections.
5. Use the down navigation button to select **Setup** and press the button.
6. Select **Algorithm** and press the button.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want from the **Favorite**, **Mono or Stereo** menus.

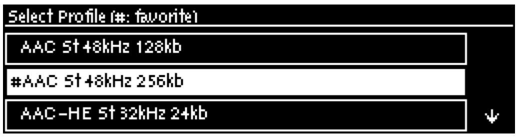


	Features	Codec Home Screen Elements
1	Favorite	Displays a list of favorite profiles that have been selected manually within the codec by users
2	Mono	Displays preprogrammed mono profiles within the codec
3	Stereo	Displays preprogrammed stereo profiles within the codec

Adding a Profile into the Favorite Menu

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Algorithm**.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want from the **Mono or Stereo** menus.
9. Press the hatch (pound) button to add the profile into the **Favorite** menu.

Profiles that have been added into the **Favorite** menu are identified by the hatch (pound) symbol next to their name after they have been selected.



Deleting a Profile from the Favorite Menu

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Press the button to select **Algorithm**.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want to delete from the **Favorite** menus.
9. Press the hatch (pound) button to delete the selected profile from the favorite menu.

17.17 Merlin Algorithm Profiles

The following algorithm profiles are programmed into Merlin codecs.

	Profiles			
	Algorithm	**Mono/Stereo**	**Sample Rate (kHz)**	**Bit rate (kbps)**
1	AAC	Mono	48	64
2	AAC	Stereo	48	128
3	AAC	Stereo	48	256
4	HE-AAC	Mono	32	16
5	HE-AAC	Stereo	32	24
6	HE-AAC	Stereo	32	48
7	AAC-LD	Mono	32	48
8	AAC-LD	Stereo	32	64
9	AAC-ELD	Mono	32	24
10	AAC-ELD	Stereo	32	48
11	aptX Enhanced	Mono	32 (16 bit)	128
12	aptX Enhanced	Mono	48 (24 bit)	288
13	aptX Enhanced	Stereo	32 (16 bit)	256
14	aptX Enhanced	Stereo	48 (24 bit)	576
15	G.711	Mono	8	64
16	G.722	Mono	16	64
17	MPEG 1 Layer 2	J-Stereo	32	128
18	MPEG 1 Layer 2	J-Stereo	48	192
19	MPEG 1 Layer 2	Mono	24	64
20	MPEG 1 Layer 2	Mono	48	256
21	MPEG 1 Layer 2	Stereo	32	128
22	MPEG 1 Layer 2	Stereo	48	256
23	Music	Mono	32	28.8
24	Music	Mono	32	48
25	Music	Stereo	32	64
26	Music	Stereo	32	96
27	MusicPLUS	Mono	48	48
28	MusicPLUS	Mono	48	96
29	MusicPLUS	Stereo	48	96
30	MusicPLUS	Stereo	48	128
31	MusicPLUS	Stereo	48	192
32	PCM Mono	Mono	48 (16bit)	768
33	PCM Stereo	Stereo	48 (16bit)	1,540
34	Opus Mono	Mono	48	64
35	Opus Stereo	Stereo	48	128

17.18 Merlin Backup Options

Tieline codecs feature highly advanced backup and redundancy options to maintain reliable audio codec streaming. These include the options outlined in the following table:

Tieline Audio Codec Backup Features			
Backup Option	**Transport: IP, ISDN or POTS**	**Time Required to Respond**	**How to Enable**
SmartStream PLUS	IP Only (Note: 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	IP Only: 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 (cold) Failover	All transports (Note: 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
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

* Note: POTS can take up to 60 seconds to connect successfully.

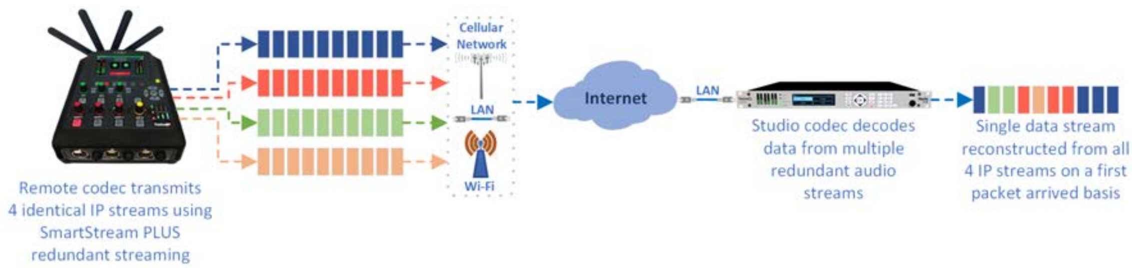
Important Note: Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.

SmartStream PLUS Redundant IP Streaming

Tieline's proprietary SmartStream PLUS IP technology ensures you're always on the air. The codec features dual Ethernet IP ports allowing two completely independent IP connections. There are three levels to SmartStream PLUS IP streaming.

1. The codec can stream simultaneous redundant data streams from both Ethernet ports and deliver seamless 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 two 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.

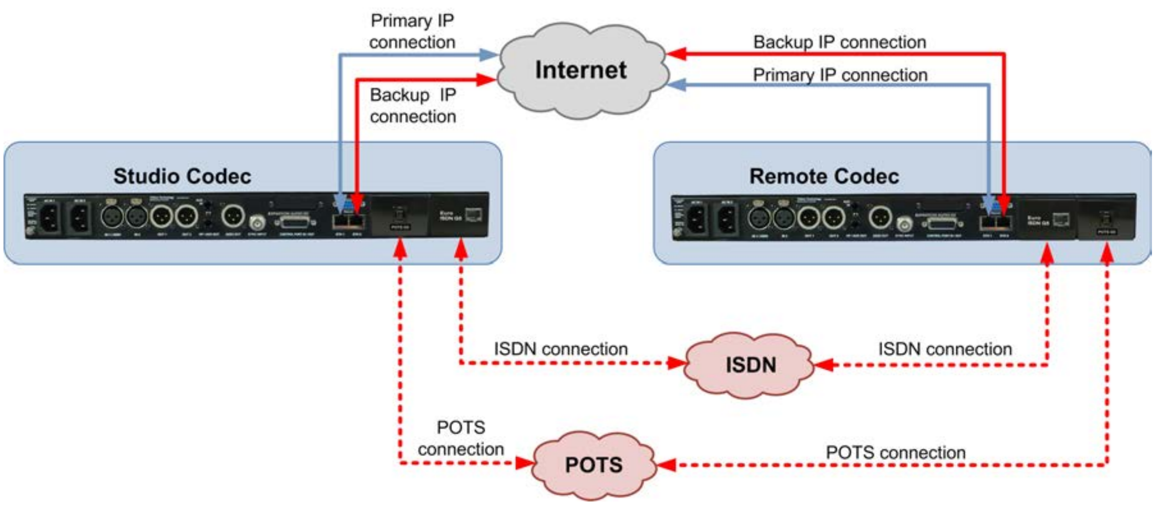
These combined measures ensure Tieline is capable of offering a rock solid IP audio solution for distributing IP audio economically and efficiently across broadcast networks. See the procedures for configuring different programs using the web-GUI for more configuration details.



On-Demand Failover (IP, ISDN or POTS)

On-demand failover requires configuration of a primary connection and an on-demand 'cold' 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.

The codec can be configured to switch to a backup connection over IP, ISDN or POTS as required. For example, you can create a program with IP as the primary connection and also create a backup ISDN or POTS connection in the same program. For details on configuring backup connections using failover see Configuring Merlin Point-to-Point Programs.



Forward Error Correction (FEC)

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 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.

Auto reconnect can be enabled when configuring a codec program designed to dial another codec or codecs. See the procedures for configuring different programs using the web-GUI for more configuration details.

17.19 Lock or Unlock a Program in the Codec

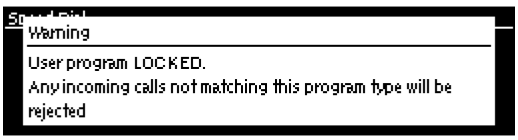
It is possible to lock a loaded custom program in a codec to ensure the currently loaded program type, e.g. mono, cannot be unloaded by a codec dialing in with a different program type, e.g. stereo.

For example, if your routing requirements require the codec at the studio to always connect using mono peer-to-peer plus IFB, simply load and lock this program in the codec. Another reason for locking a program in the answering codec is to always use a particular jitter buffer or FEC setting.

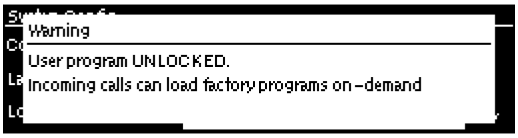
Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will be rejected. A compatible program type can still connect and specify different connection parameters such as algorithm preferences and bit rates via 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 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. 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.

17.20 Locking the Front Panel

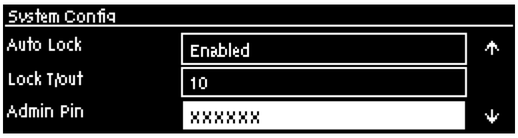
The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default.

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 down to each in turn and press to enter a new PIN.

18 Connecting to the ToolBox Web-GUI

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

1. HTML5 Toolbox Web-GUI: fully configure codec settings, create dialing programs and dial, hangup and monitor connections.
2. HTML5 Toolbox Quick Connect Web-GUI: designed for dialing and managing simple peer-to-peer connections; ideal for non-technical users.

About the HTML5 Toolbox Web-GUI

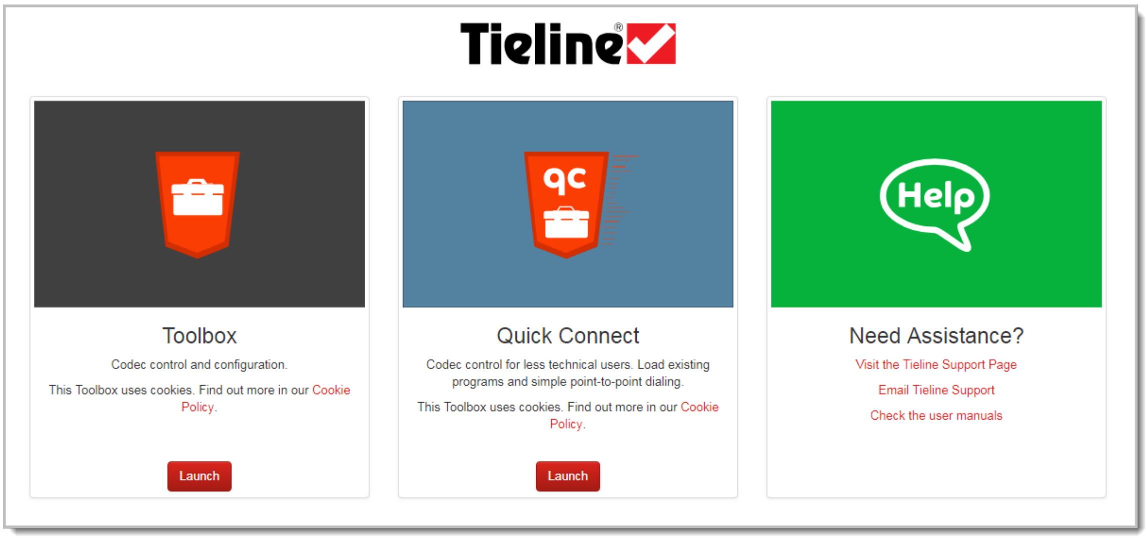
The HTML5 Toolbox Web-GUI improves the user experience with G5 codec command and control and runs seamlessly on modern browsers. The HTML5 Toolbox Web-GUI will run on Mac, Windows and Linux computers.

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 POTS, ISDN or IP. To enable the **Quick Connect panel** press the **SETTINGS** button, then navigate to **WebGUI > Quick Connect > Enabled**.

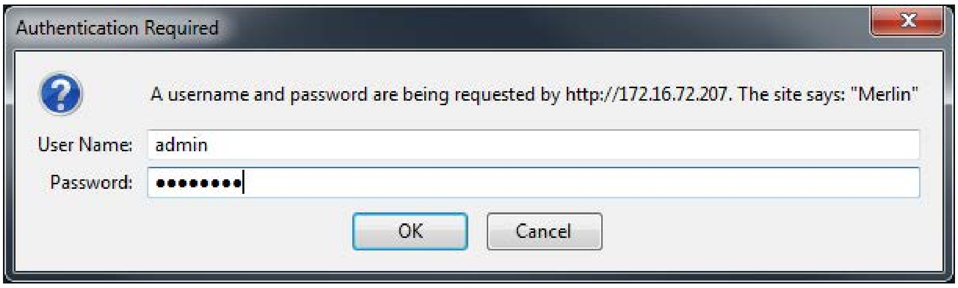
18.1 Opening the HTML5 Web-GUI & Login

1. Attach an Ethernet cable to the **ETH1** 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.



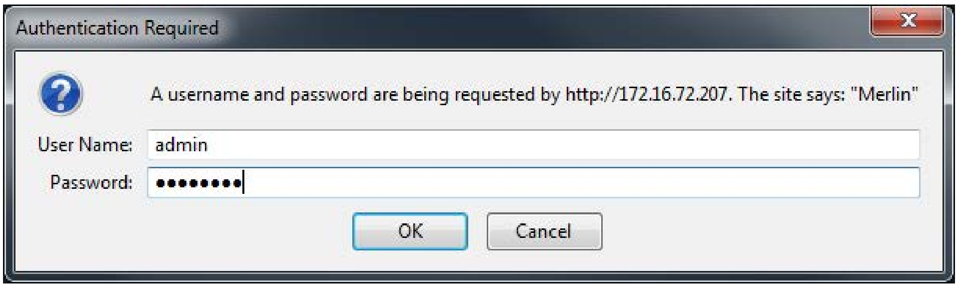
Launching the HTML5 Toolbox Web-GUI

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 user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline **highly recommends** you change the password (see Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.



Launching the HTML5 Toolbox Quick Connect

1. Click to launch the **HTML5 Toolbox Quick Connect** Web-GUI.
2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline **highly recommends** you change the password (see Changing the Default Password). This will provide 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 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 button to select **WebGUI** and press the  button.
3. Navigate to **Browser Title** to enter the custom title.
4. Use the on-screen keyboard to enter the title, then navigate to **Enter** in the bottom right-hand corner of the **SCREEN** and press the  button.

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**.

LAN Troubleshooting

PC LAN Settings

Check the LAN settings on your PC if it is connected to a LAN and is having trouble opening the Toolbox Web-GUI in a web-browser.

1. Open Internet Explorer.
2. Click **Tools > Internet Options > Connections**.
3. Click the **LAN settings** button.
4. If the PC is using a proxy server over the LAN you may need to select the **Bypass proxy server for local addresses** option box.
5. If you still can't connect, click the **Advanced** button in the **LAN Settings** dialog and ask your IT administrator to assist you with entering the IP address of the codec into the **Exceptions** pane of the **Proxy Settings** dialog.

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 button to navigate down to **WebGUI** and press the button.
3. Select **Alt. 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 IP address of your codec into your browser with a full colon and then the new port number.

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

18.2 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

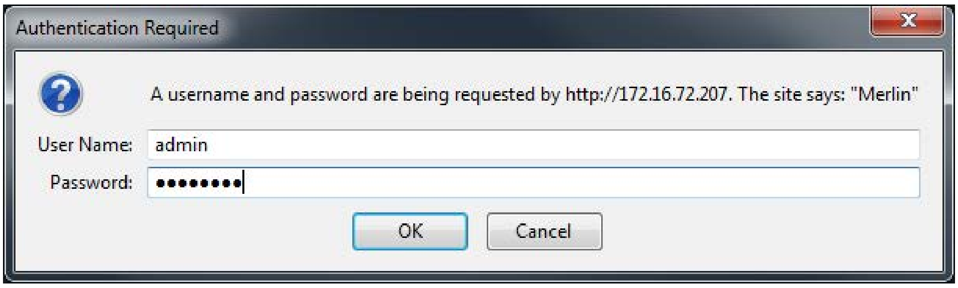
Adequate security is a major factor in ensuring your codecs and your broadcast network remain secure. There are several layers of security available in Tieline codecs to maintain secure connections. These include:

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 White and Blacklists 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 to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Adjust settings using the **Options panel** in the Toolbox HTML5 Web-GUI, 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-GU, 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.

Changing the Default Password

The default password for the Toolbox Web-GUI is **password**. Enter this in the authentication dialog to use the Web-GUI initially and then Tieline highly recommends changing the default password to protect your codec from being tampered with during live broadcasts. Note: In the HTML5 Web-GUI authentication dialog it is necessary to enter **admin** as the **User Name**.



Toolbox HTML5 Web-GUI Login Dialog on a Merlin Codec

Creating a New Password

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

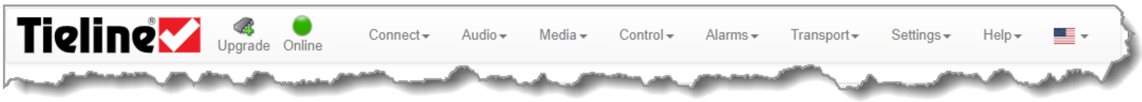
1. Press the **SETTINGS** button.
2. Use the navigation button to select **WebGUI** and press the button.
3. Select **Password** and press .
4. Use the **KEYPAD** to enter a new password and press the button to save the new setting (Note: there is no character limit for passwords).

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.

19 Using the HTML5 Toolbox Web-GUI

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 logging in subsequently. The default panel view is displayed on login if cookies have been cleared.

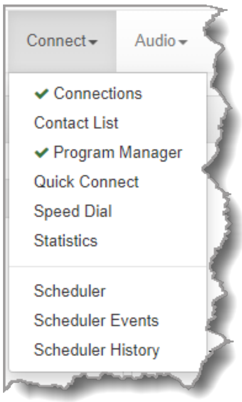
The green **Online** indication in the top left-hand corner of the Toolbox Web-GUI indicates it is online and can be used for codec control. A red **Offline** indication is displayed when the codec is unavailable. The **Upgrade** symbol is displayed when a new firmware version is available for the codec. Open the **Firmware panel** in the **Settings** menu to upgrade the codec with new firmware.

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 Size or Screen Position

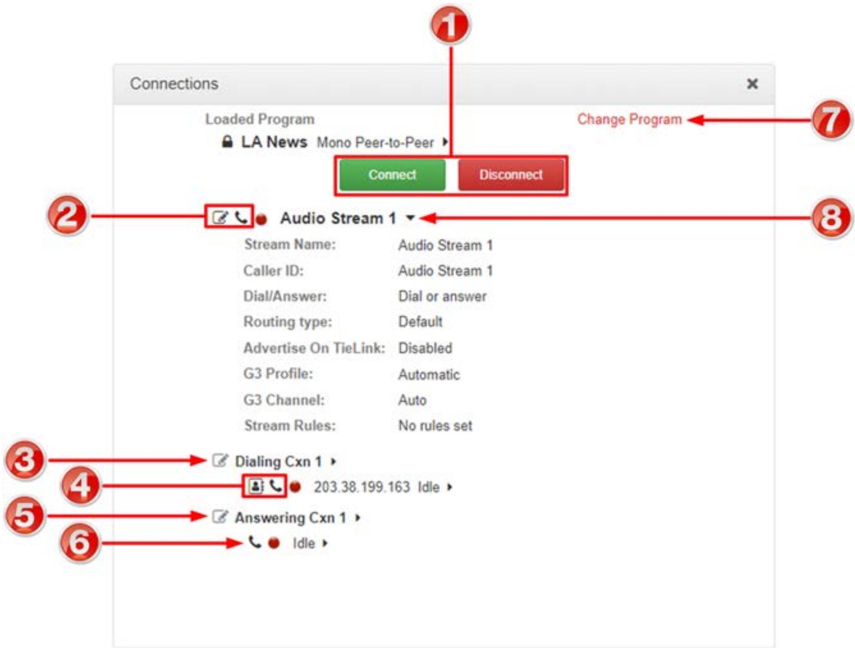
Click an item in the **Menu bar** to display available panel options, then click to select and open a panel. New panels automatically open in the top left of the screen. A green **Tick** adjacent to a panel name in the menu signifies it is already open in the web-GUI.



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.

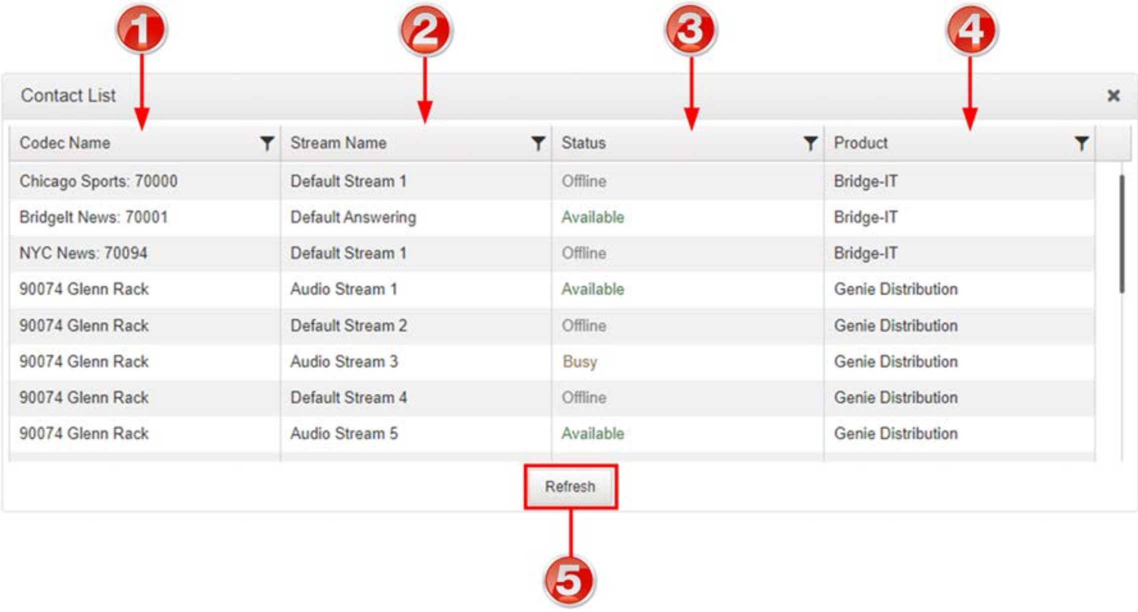
Connect Panels: Load & Connect Programs & Manage Audio Streams

Connections Panel



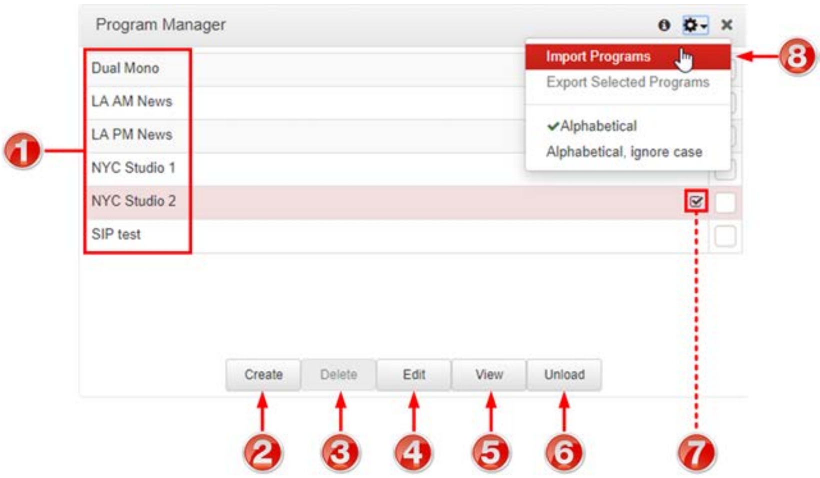
	Feature	Description
1	Program **Connect / Disconnect** button	Click to connect/disconnect all audio streams in a program.
2	Audio Stream **Edit** and **Connect / Disconnect** buttons	Click the **Edit** button to edit audio stream settings. Click the **Connect/Disconnect** button to connect/disconnect all connections in an audio stream.
3	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.
4	TieLink **Address Book** and Connection **Connect / Disconnect** button	Click the **Connect/Disconnect** button to connect/disconnect an individual connection. Also adjust the connection 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.
5	Answering connection **Edit** button	Click to edit answering connection settings
6	Answering Connection **Disconnect** button	Click to disconnect an answering connection
7	**Change Program**	Click to unload the current program and load a new program.
8	Show/Hide Arrow	Click to show/hide audio stream and connection details.

Contact List



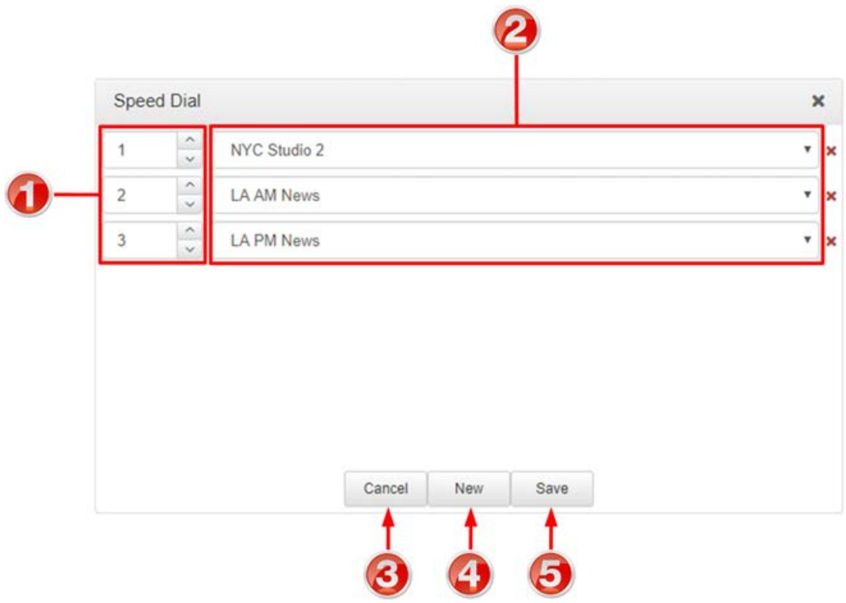
	Feature	Description
1	Codec Name	The name given to a codec to identify it when a program is created
2	Stream Name	The name given to a stream to identify it when a program is created
3	Status	Displays the status of an audio stream, which can be **Available**, **Busy** or **Offline**
4	Product	Displays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

Program Manager Panel



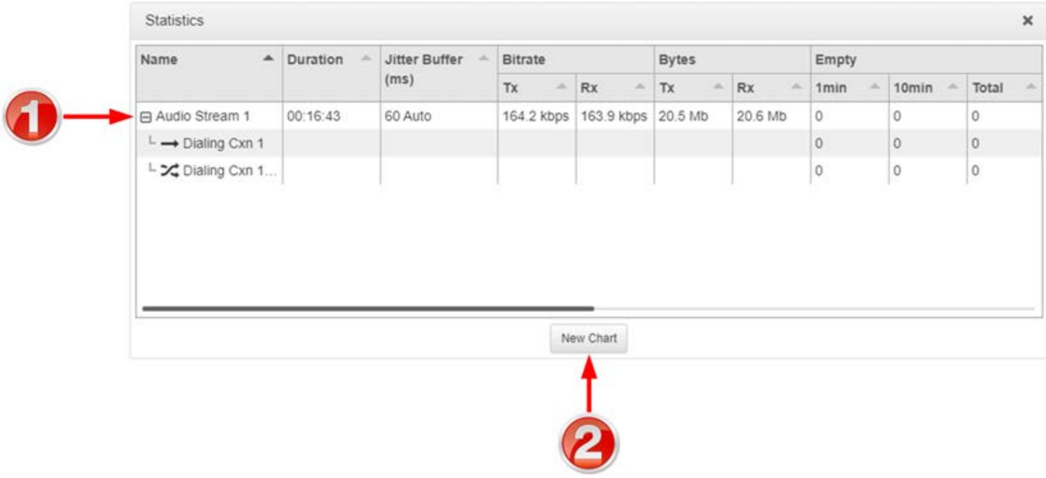
	Feature	Description
1	Program list	The list of saved programs in the codec
2	Create New Program button	Click to create a new program using the program wizard.
3	Delete Selected Programs button	Click to delete all selected programs
4	Edit Selected Program button	Click to edit the selected program
5	View 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



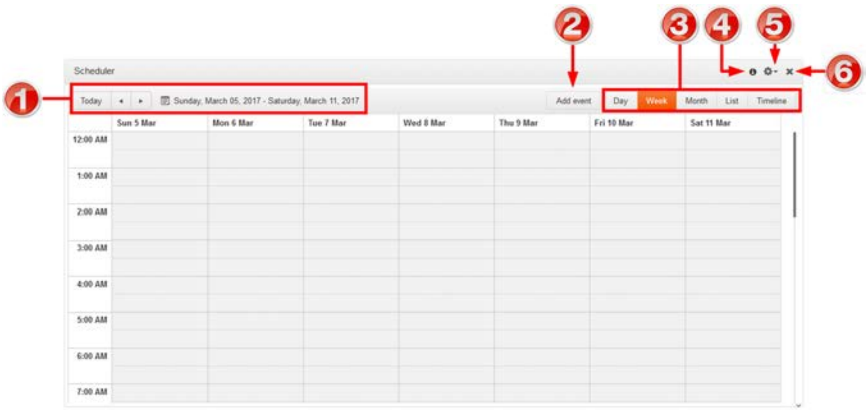
	Feature	Description
1	Speed dial numbers	The number allocated for each speed dial configured
2	Speed dial program	Select the program to associate with a speed dial number
3	**Cancel** button	Click to cancel pending changes
4	**New** button	Click to create a new speed dial
5	**Save** button	Click to save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

Statistics Panel



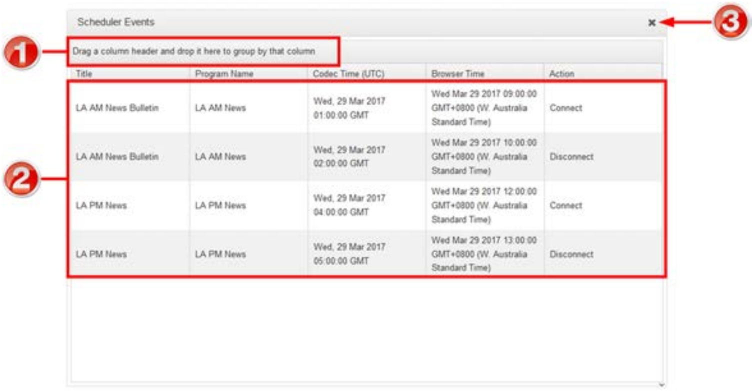
	Feature	Description
1	Expand/Collapse	Click to show/hide audio stream statistics, including packet arrival data info.
2	**New Chart** button	Click the **New Chart** button to select a **Data Series** and create a customized **Statistics panel**.

Scheduler Panel



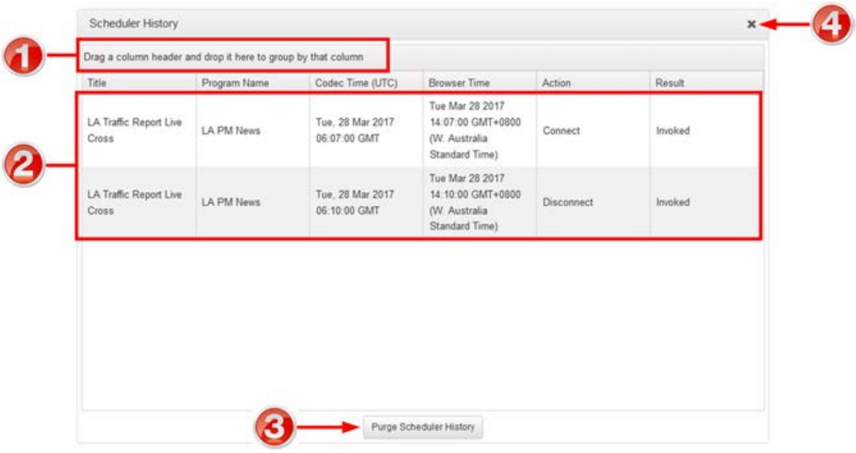
	Feature	Description
1	Date selection	Select the days you wish to view in the scheduler.
2	Add event button	Click to create a new scheduled event.
3	View type	Click to select the timeframe or timeline view of scheduled events.
4	**Information symbol**	Hover over the **Information symbol** to view **Scheduler panel** information.
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.

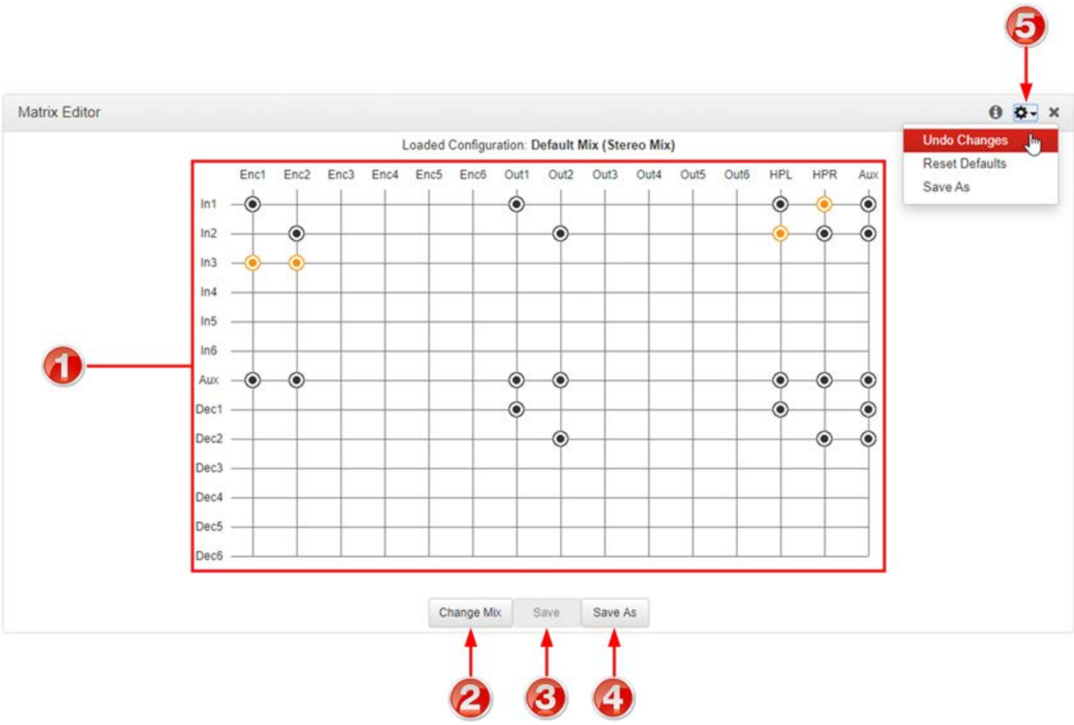
Audio Menu Panels

Inputs Panel

Important Note: Tieline codecs have different input configurations, therefore the image shown may not reflect the number of inputs displayed in your codec Web-GUI.

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

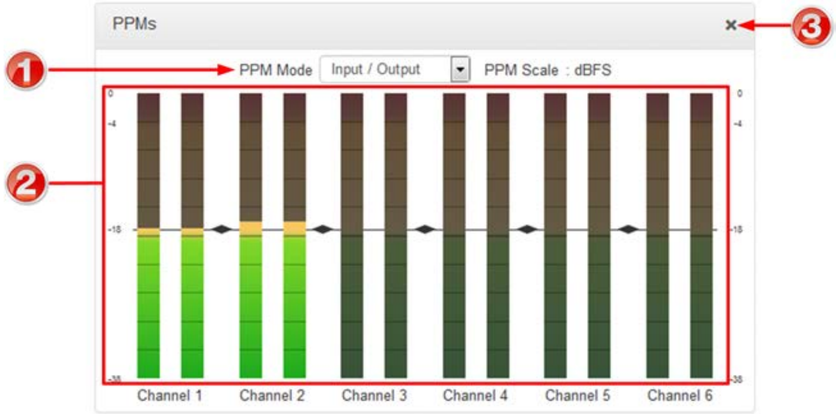
Matrix Editor Panel



	Feature	Description
1	Matrix Editor crosspoint matrix	Click to select and deselect audio crosspoints to customize routing.
2	**Change Mix**	Click to select the default or Custom matrix mix to view and edit. Options include **Load**, **Delete** and **Rename** a mix
3	**Save**	Click to save changes to a custom matrix mix
4	**Save as**	Click to save as a new custom matrix mix
5	**Options menu**	Menu options available to undo any changes, reset the mix to the factory default matrix mix for the program, or save as a new mix.

PPMs Panel

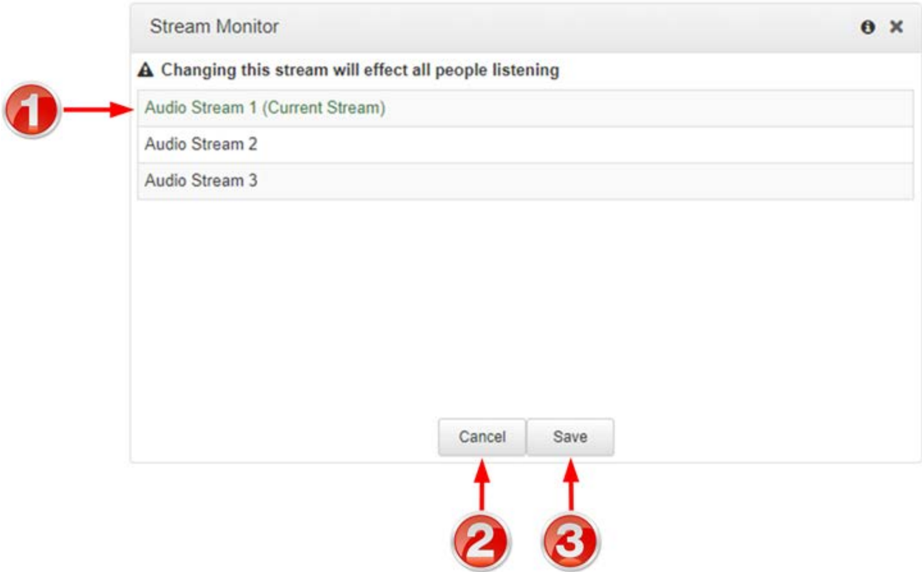
Note: Click and drag the bottom right-hand corner to expand the panel in **Input / Output** mode.



	Feature	Description
1	**PPM Mode**	Select a preferred PPM meter mode. Options include **Program Default**, **Input**, **Decoder**, **HP Monitor** and **Input / Output**
2	PPM Meters	6 PPM meters using dBFS metering
3	**Close** button	Click to close the panel.

Important Note: Codec front panel **PPMs 1-6** and PPMs in the **PPMs panel** always reflect the default program settings and do not display customized mix adjustments using the **Matrix Editor**.

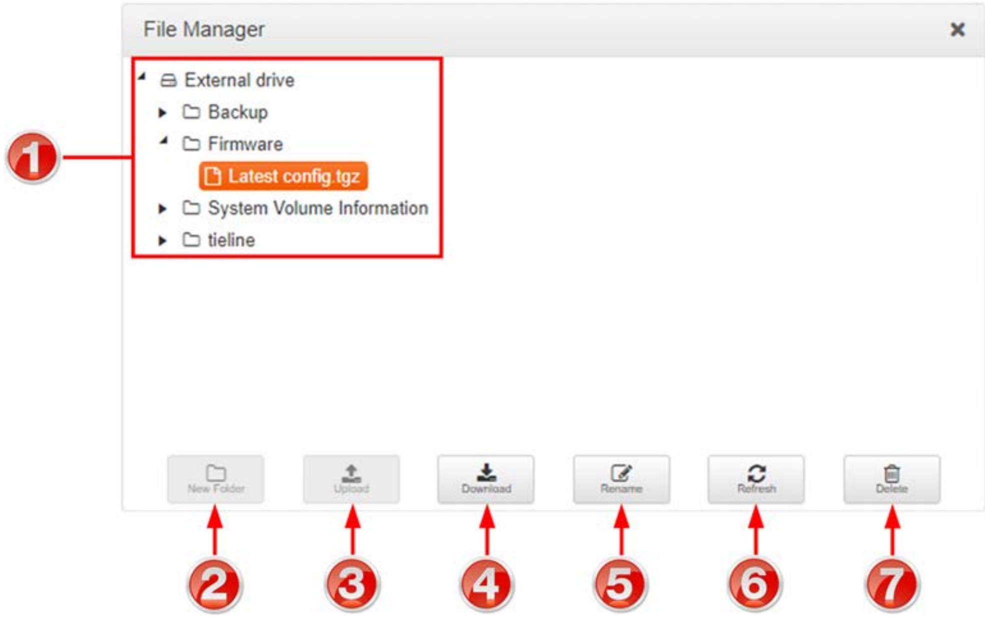
Stream Monitor



	Feature	Description
1	Current Stream	Currently selected stream being monitored
2	**Cancel** button	Cancel selection of a new stream to monitor
3	**Save** button	Click to save a selected stream as the new stream to monitor

Media Menu

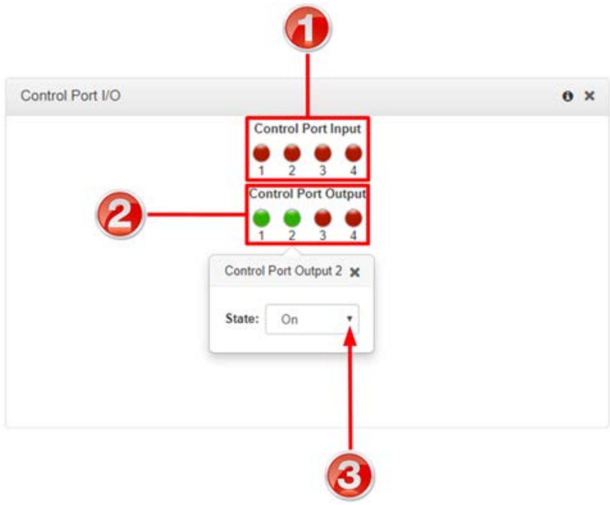
File Manager



	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

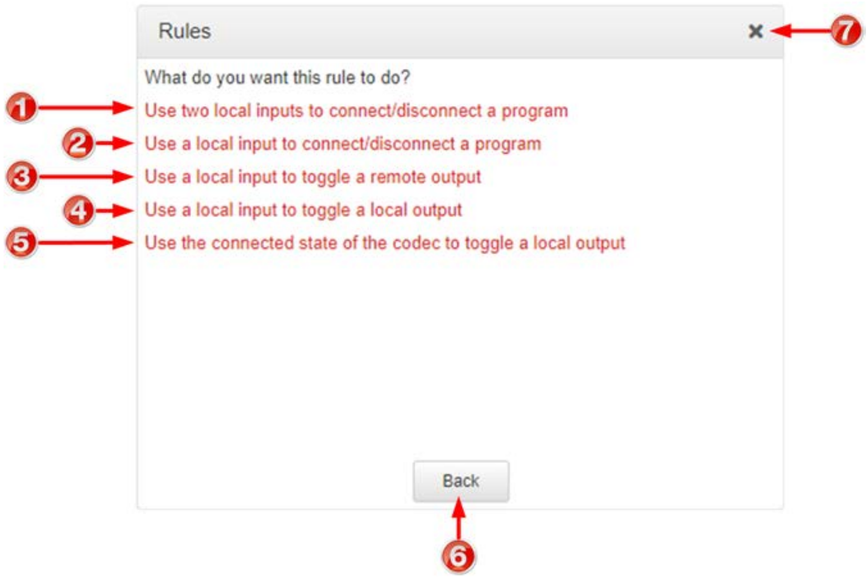
Control Menu

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

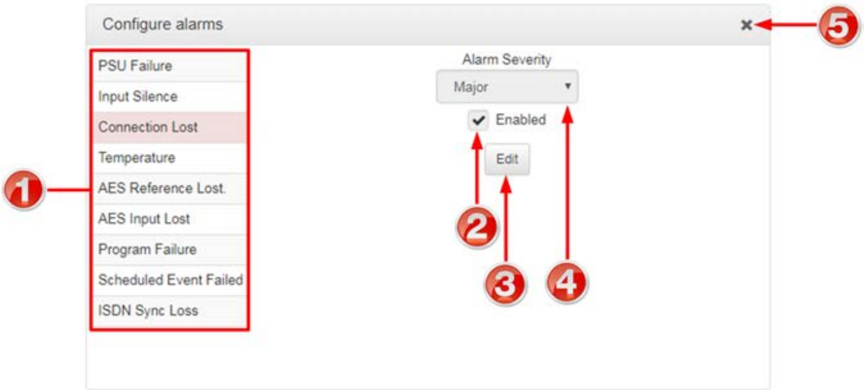


	Rule	Description
1	Use two local inputs to connect/disconnect a program	Click to configure connection and disconnection after different relay inputs are switched **ON**.
2	Use a local input to connect/disconnect a program	Click to configure connection and disconnection by toggling an input.
3	Use a local 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	**Back** / **Add New Rule** button	Click to add a new rule, or exit the rule creation function.
7	**Close** button	Click to close the panel.

Alarms Panels: Configure & Monitor Alarms

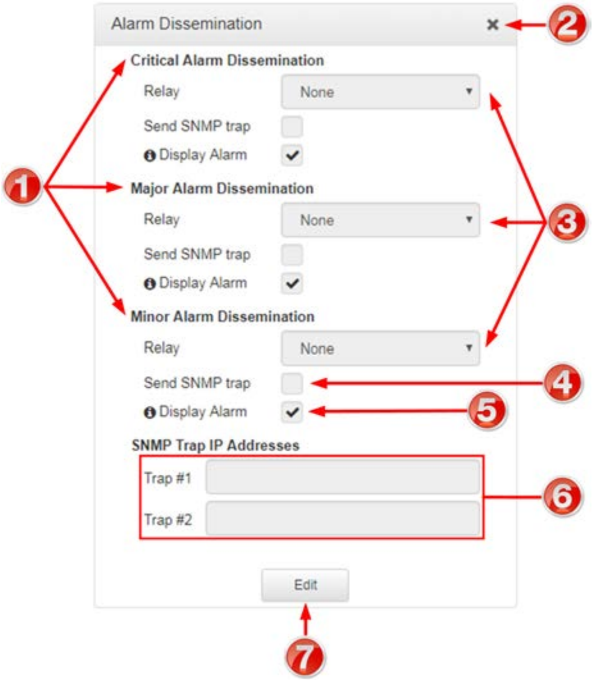
Configure Alarms Panel



	Feature	Description
1	List of alarm types	Click to select an alarm type to configure.
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

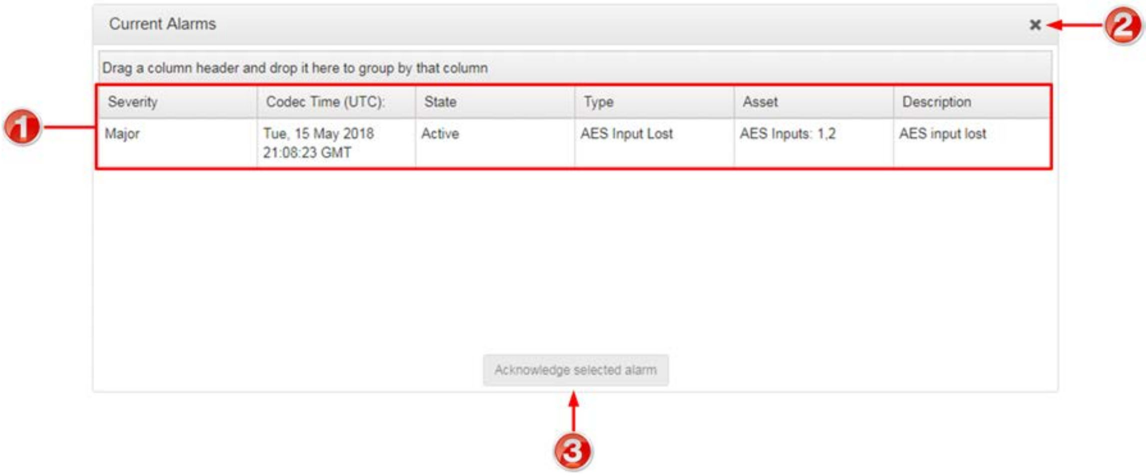
Alarm Dissemination Panel

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



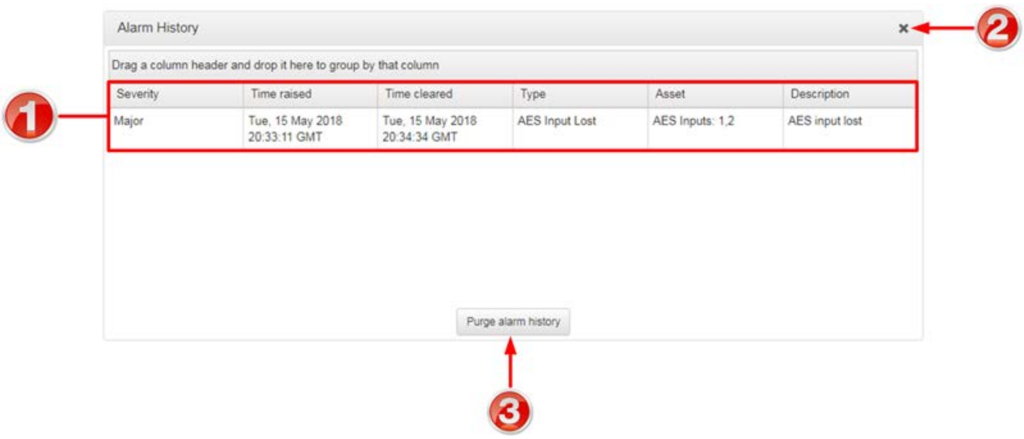
	Feature	Description
1	List of alarm severity levels	Click to select an alarm severity level to configure it.
2	**Front panel alarm & Toolbox** check-box	Select the check-box (default enabled) to deliver front panel **ALARM LED** notifications and HTML5 Toolbox Web-GUI alarm notifications.
3	**Close**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

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.

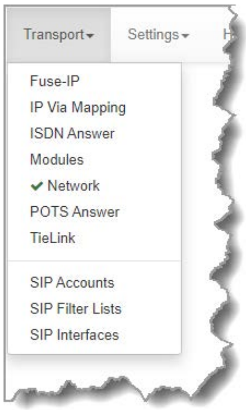
Alarm History



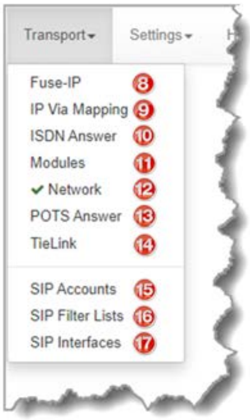
	Feature	Description
1	Alarm history description	View the history of previous alarms in the codec.
2	**Close** button	Click to close the **Alarms** panel.
3	**Purge Alarm History** button	Click to clear the alarm history.

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 panels is also provided.



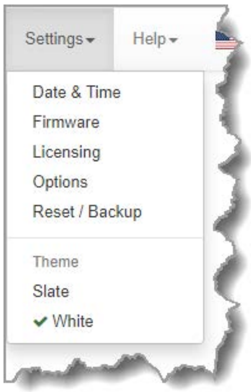
	Feature	Description
1	Network interfaces	Click to select and edit, or view network configuration settings for each Ethernet and VLAN interface.
2	**Details** tab	Display configuration options for a selected network interface, plus other device details.
3	Control/Streaming	Select **Control** and/or **Streaming** options for the selected interface.
4	**Link Mode**	Configure the Ethernet, VLAN or Wi-Fi 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	**Save/Undo** button	Click **Save** to store settings, or click **Undo** to revert to previously configured settings.
8	**Fuse-IP**	Click to open the **Fuse-IP panel** and configure Fuse-IP bonding
9	**IP Via Mapping**	Configure default **Primary**, **Secondary** and **Tertiary** interfaces.
10	**ISDN Answer**	Click to open the panel and configure ISDN Answering settings.
11	**Modules**	Click to edit hardware module configuration.

12	**Network**	Click to open the **Network panel** and configure network settings.
13	**POTS Answer**	Click to open the panel and configure POTS Answering settings.
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 Whitelist**. Add SIP URIs to the **URI Blacklist** and add user agents to the **User Agent Blacklist** to deny them access to the codec.
17	**SIP Interfaces**	Click to open the panel and configure port, proxy and **Via** 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 different codec configuration settings and options. Click to select and open each panel. It is also possible to change the HTML5 Web-GUI theme.

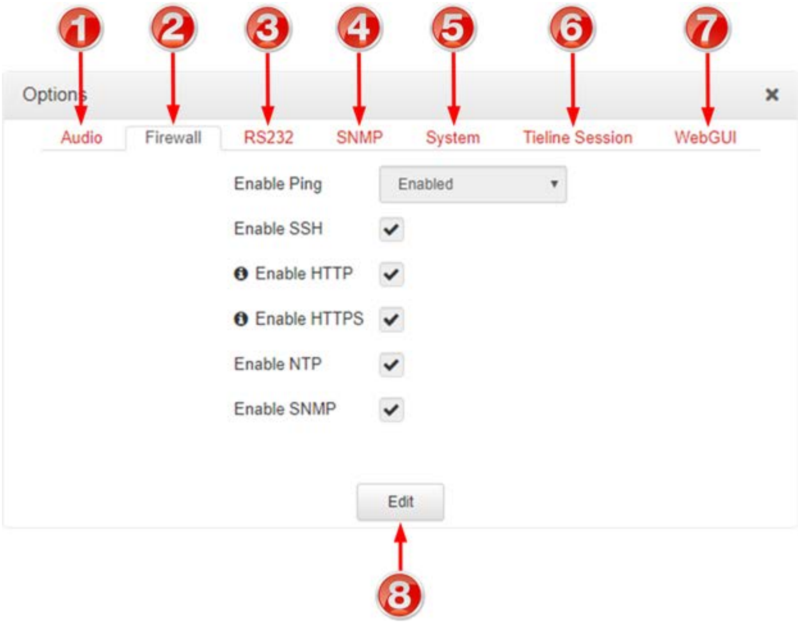
As an example, the **Options panel** is displayed with a brief description of the other panels available.



Settings panels

	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, firewall, 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	**Theme**	Adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI; options include **White** or **Slate**.

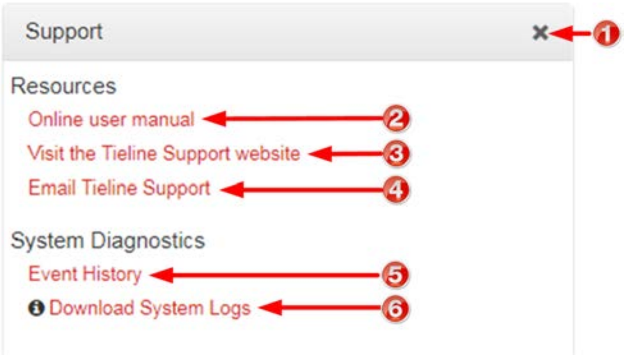
Options panel



	Feature	Description
1	**Audio** Settings	Configure various audio settings on the codec, including analog input PPM units, reference levels and phantom power voltage.
2	**Firewall** Settings	Enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.
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 (Note: the codec name displayed will vary by product type)
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

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.

	Feature	Description
1	**Close** button	Click to close the panel.

Language Selection

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

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


Help
English
French
Portuguese
Spanish

19.1 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, ISDN or POTS.

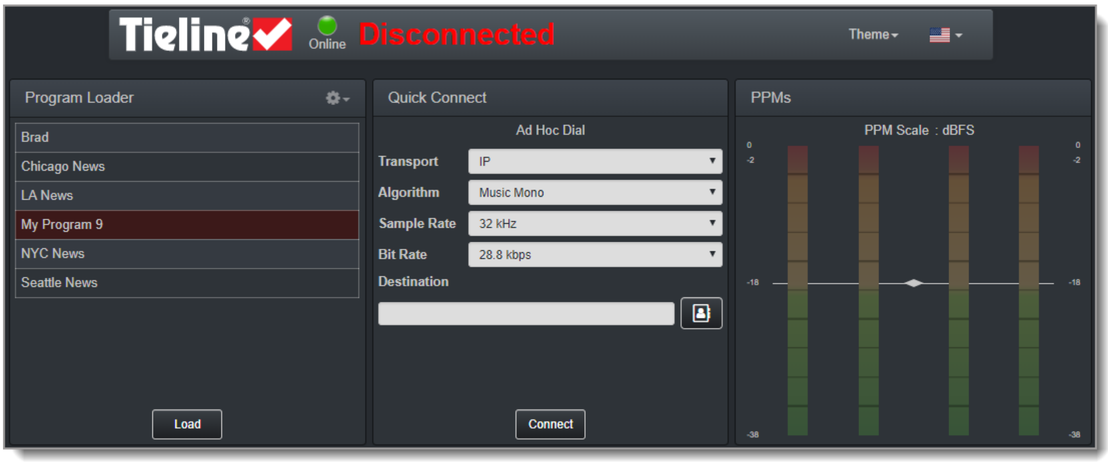
See Opening the HTML5 Web-GUI and Login for details about launching the standalone HTML5 Quick Connect Web-GUI. The **Quick Connect panel** can also be launched from the **Connect** menu in the HTML5 Toolbox Web-GUI.

Important Notes:
- 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 > Enabled**.

Launching the HTML5 Quick Connect Web-GUI

1. Type the codec IP address in 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.

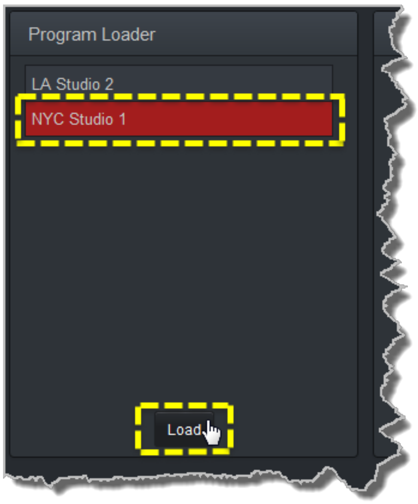


Important Note: To change the password using the codec front panel navigate to **Settings > WebGUI > Password** and press the OK button. Use the keypad to enter a new password and press the OK button to save the new setting.

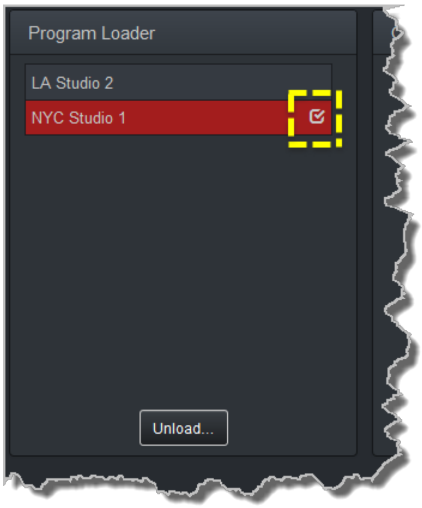
Loading and Unloading an Existing Program

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 the **Load button** to load it in the codec.



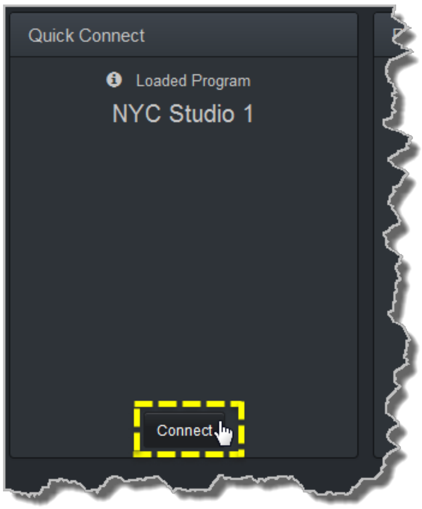
2. The **Check-box symbol** appears next to the program name to confirm it has been loaded and the **Load button** changes to an **Unload button**.



To unload a program click the **Unload button**.

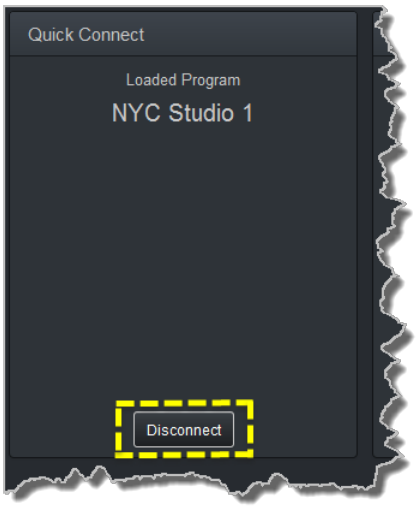
Dial a Loaded Program

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**.

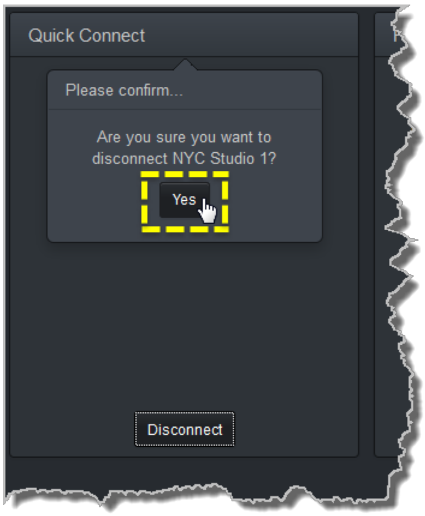


Disconnect a Loaded Program

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



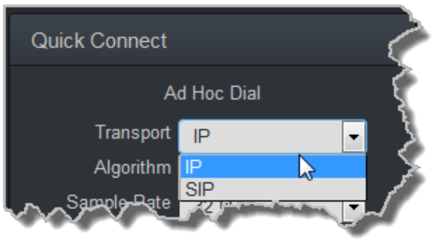
2. Click **Yes** in the confirmation dialog to disconnect the connection.



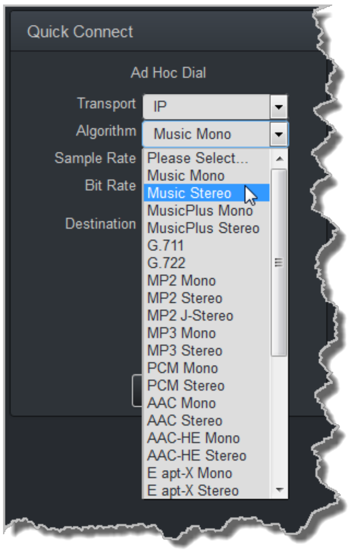
Dial Peer-to-Peer over IP with Quick Connect

Important Notes:
- Click the **Unload button** in the **Program Loader panel** if a program is currently loaded.
- The transcriptor algorithm is for closed captioning and not normal broadcast configurations.

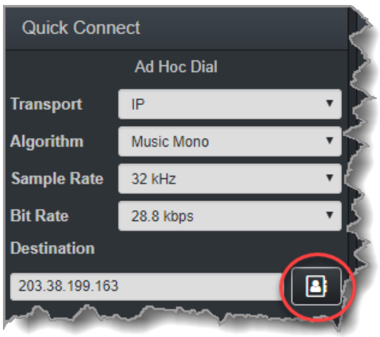
1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **IP**.



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



3. Click the 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.

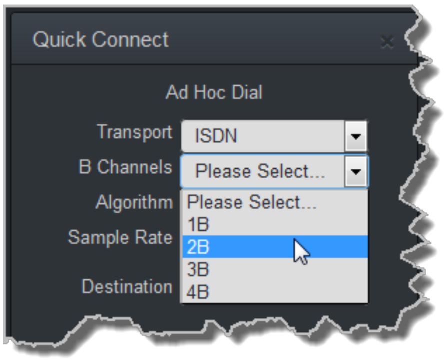


5. Click the **Connect button** to dial.

Dial Peer-to-Peer over ISDN with Quick Connect

Important Notes:
- Click the **Unload button** in the **Program Loader panel** if a program is currently loaded.
- The transcriptor algorithm is for closed captioning and not normal broadcast configurations.

1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **ISDN**.
2. Click the drop-down **Algorithm** menu and select an algorithm.
3. Click the drop-down **B Channels** menu and select the number of channels required for this connection.

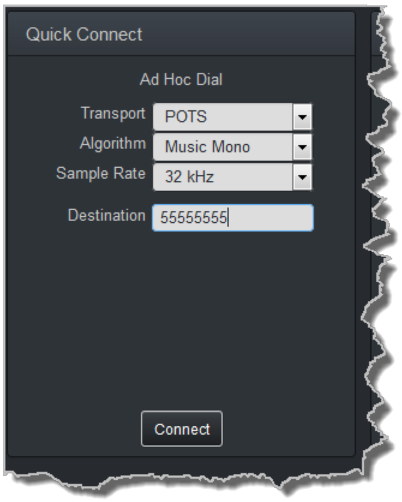


4. Click to select the appropriate **Sample Rate** for the connection.
5. Click in each **Destination** text box in turn to enter the ISDN number for each B Channel.
6. Click the **Connect button** to dial.

Dial Peer-to-Peer Over POTS with Quick Connect

Important Notes:
- Click the **Unload button** in the **Program Loader panel** if a program is currently loaded.
- The transcriptor algorithm is for closed captioning and not normal broadcast configurations.

1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **POTS**.
2. Click the drop-down **Algorithm** menu and select an algorithm. The connection bit rate is configured automatically. Note: G5 POTS modems initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.
3. Enter the phone number in the **Destination** text box.



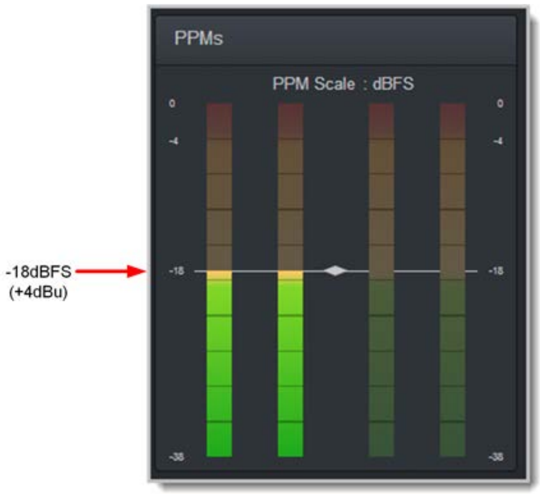
4. Click the **Connect button** to dial.

Monitoring PPMs

Set audio levels so that audio peaks average at the nominal 0vu point indicated below on the PPM meters. 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.

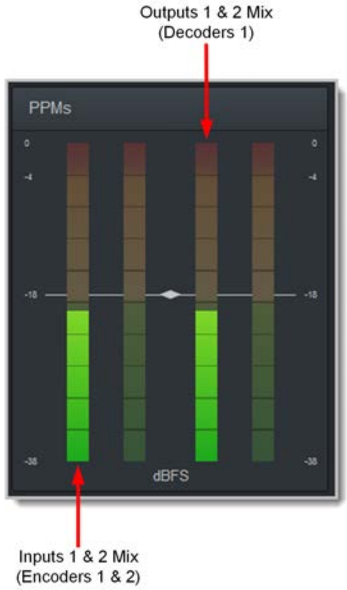
The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session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

The default Tieline G5 audio reference scale displayed on the PPMs is -38dBFS to 0dBFS (e.g. Merlin, Genie, Bridge-IT and ViA codec families). Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 18dB of headroom from the nominal 0vu point.



Mono and Stereo PPM Metering

When connected with a mono program the codec will display a mix of inputs 1 and 2 on **PPM1**. **PPM 3** displays the level of return audio.



When connecting with a stereo program, the codec displays audio on **PPM1** and **2** for inputs 1 and 2 and **PPM 3** and **4** for the return audio.

19.2 Configure Programs with the Connections Panel

The **Connections panel** allows:

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

The **Connections panel** delivers the ability to edit destination settings for an audio stream when other audio streams are connected, without unloading a program. Please note: If you need to use an existing program as a template for a new program, or add a custom matrix mix to a program, please use the **Program Manager panel** to create a program.

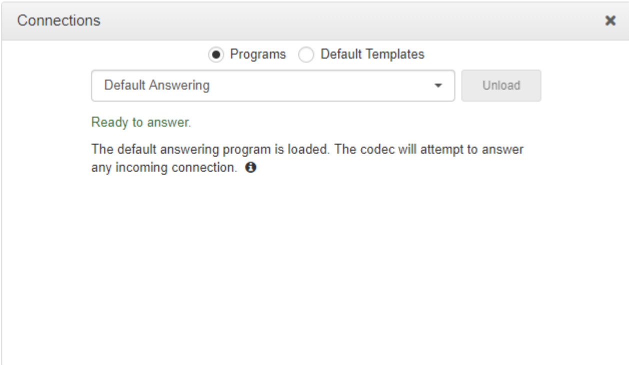
Configuring New Programs

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:

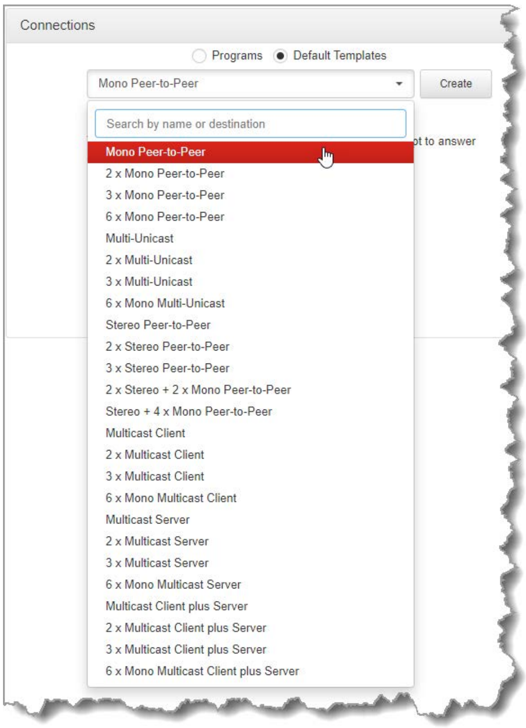
- Name or rename a program.
- Apply a custom matrix.
- Manage Program level Rules.
- Create a program from a copy of an existing program.

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

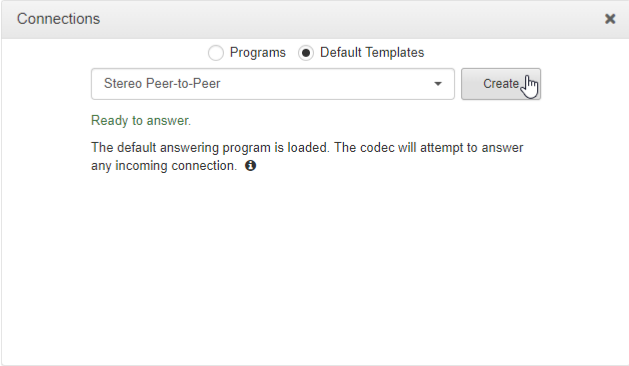
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**.



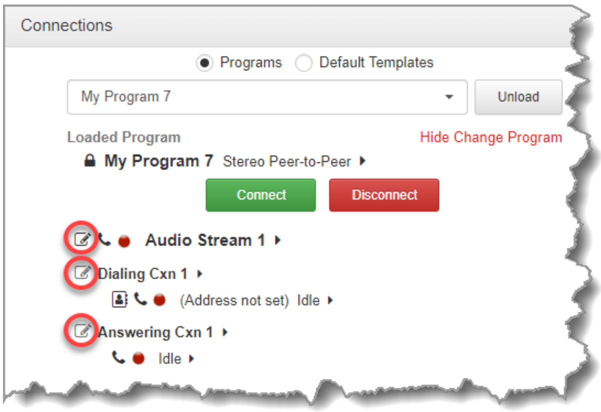
2. Select **Default Templates** and then click the drop-down arrow to select one of the supported default templates in the codec. Note: The default templates available will vary from codec-to-codec.



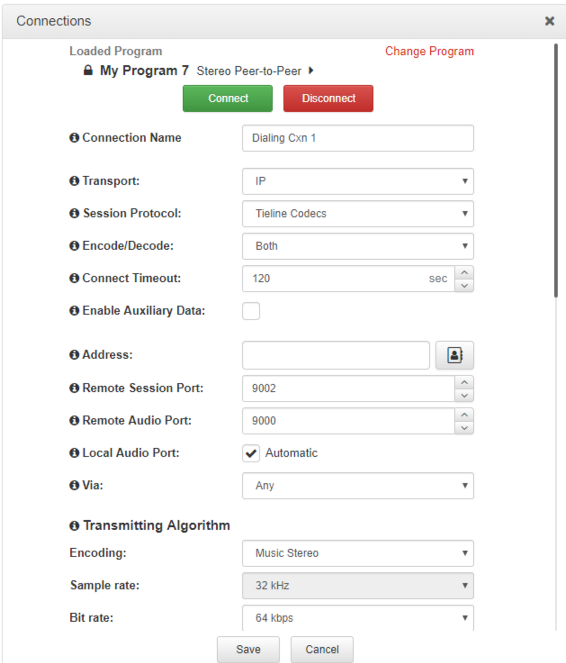
Click **Create** to add the new program.



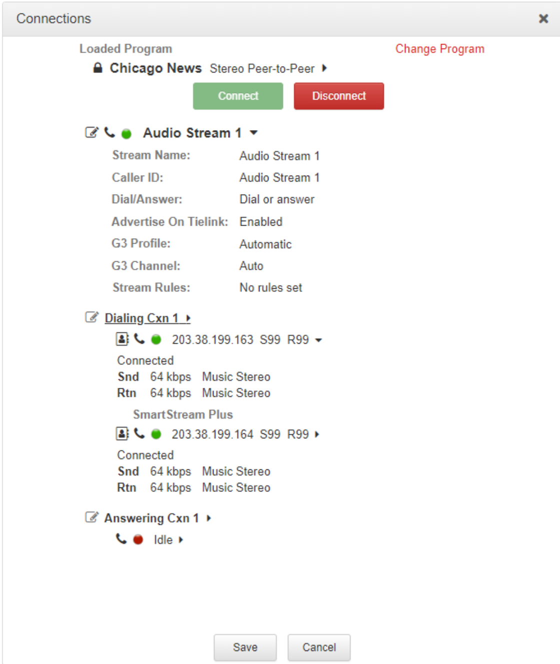
4. To configure new connection settings, or edit existing connection settings, click the **Edit symbol** .



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.



6. Connections can also be connected, disconnected and managed from within the **Connections panel**.



19.3 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 your codec 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.

If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want 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 the remote codec.

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

Configuring Ethernet Ports and VLANs

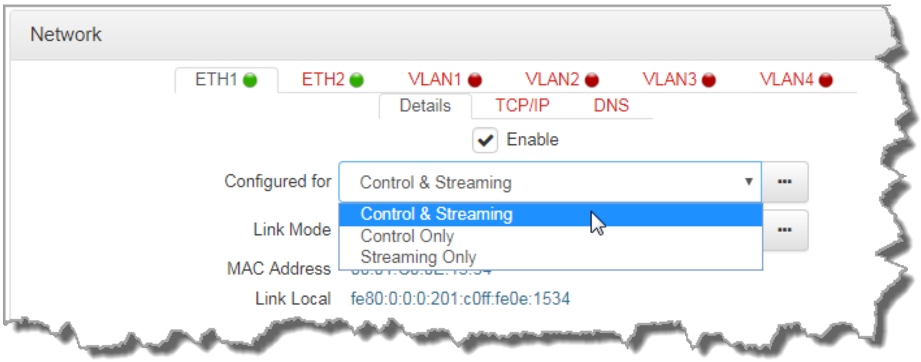
The codec features two physical Ethernet port interfaces and up to four additional VLAN interfaces.

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, 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. Ethernet and VLAN interfaces can be configured for:

- Controlling audio: codec control and command only from the Ethernet port.
- Controlling and Streaming: stream audio and control and command the codec via the Ethernet port.
- Streaming audio: stream audio only from an Ethernet port (**ETH2** and **VLANs** only).
- Nothing: Disable the Ethernet port from streaming audio and codec command and control (**VLANs** only).

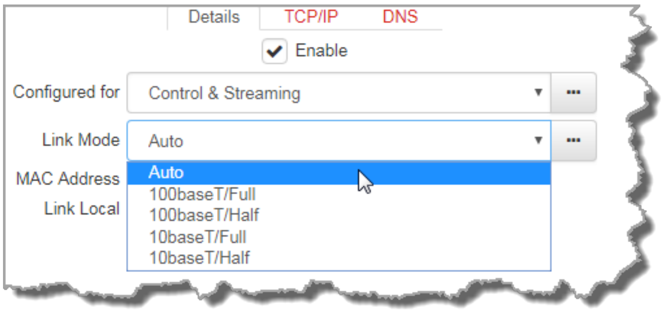
Select the **Details tab** in the **Network** panel to edit control and streaming settings. Select the **Enable** check-box to activate each interface. **ETH1** and **ETH2** are enabled by default. Note: An Ethernet interface with a green status indication is enabled and has an active LAN cable attached. An Ethernet interface with red text and a yellow status indication is enabled, however a network cable is not attached. A VLAN interface with red text and a yellow status indication is enabled but unavailable for some reason, e.g. VLAN networking is not configured correctly. An interface in red text with a red status indication is disabled, e.g. **VLANs** in the following image.



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.

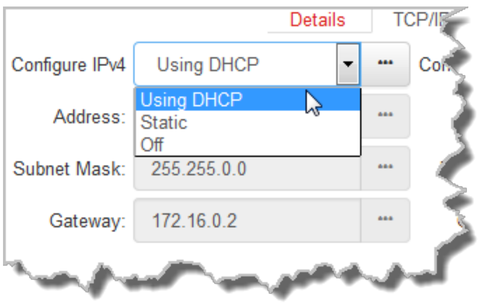
1. Click the drop-down **Link Mode** arrow and select the preferred setting.



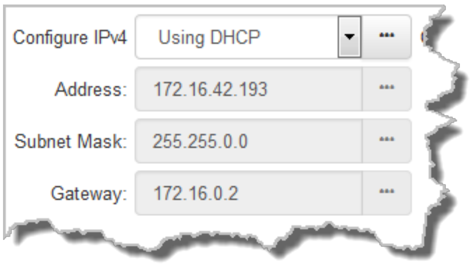
2. Click **Save** to store the new setting.

IPv4 Address Configuration

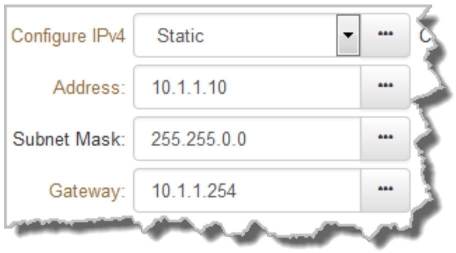
Click to select the **TCP/IP** tab in the **Network** panel to configure settings. The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the **Configure IPv4** drop-down menu. To ignore IPv4 settings select **Off**.



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.



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.

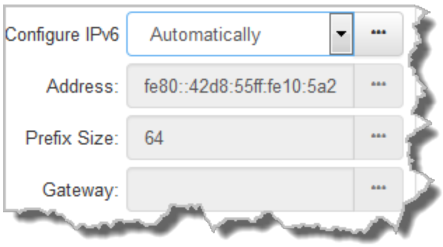
Types of IPv6 Addresses

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

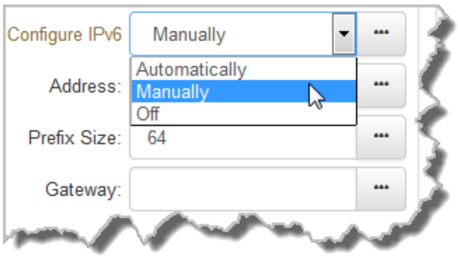
1. By default the codec is configured for connecting to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.



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

Manual IPv6 Address Assignment

1. To configure IPv6 address details into the codec manually, select **Manually** and enter details into the **Address**, **Prefix** and **Gateway** text boxes.

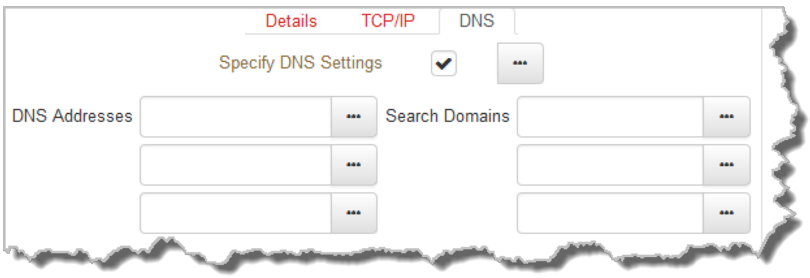


2. Click **Save** to store all configuration settings.

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**.

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



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

IP Via 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 **Via** interfaces in order of use when available are:

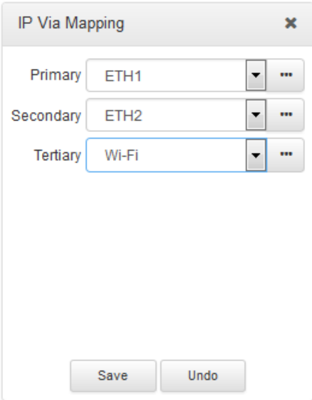
1. **ETH1** Ethernet port (default **Primary** Via interface)
2. **ETH2** Ethernet port (default **Secondary** Via interface)
3. **Wi-Fi** (default **Tertiary** Via interface)

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 Primary, Secondary and Tertiary Interfaces

It is possible to reconfigure the default **Primary** (**ETH1**), **Secondary** (**ETH2**) and **Tertiary** (**Wi-Fi**) 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 interface and for another it may be a Wi-Fi connection. This allows you to configure site-specific settings to suit the available network interfaces at different remote locations.

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



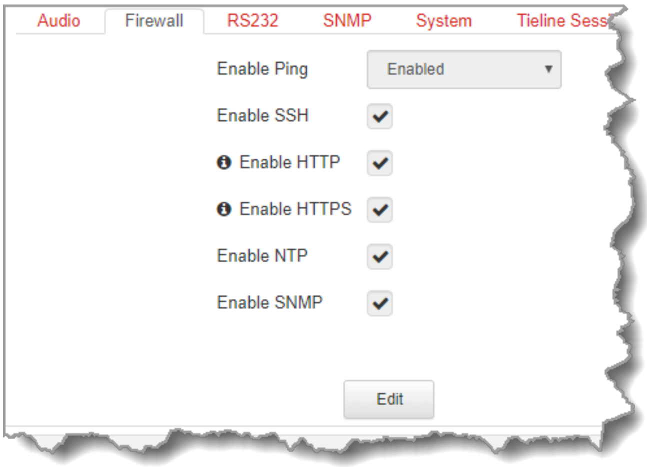
3. Click **Save** to store the configuration.

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

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.

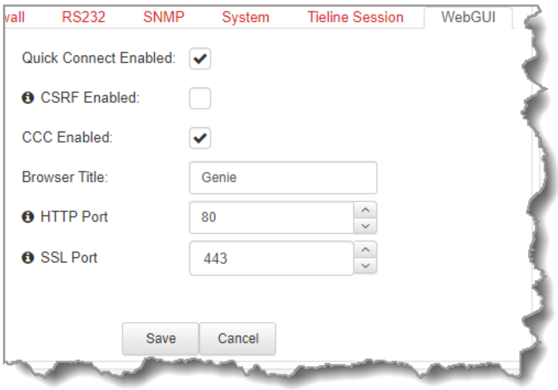
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **Firewall** and then click **Edit** to adjust settings.
3. Click **Save** to store the new configuration settings.



Configure Cross-Site Request Forgery

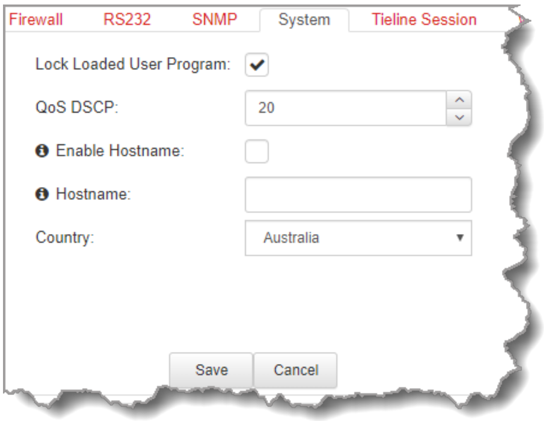
CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

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

1. Open the HTML5 Toolbox Web-GUI and click **Settings** and then click **Options** to open the **Options panel**.
2. Select **System** and then click **Edit**. Click in the **QoS DSCP** text box and enter the preferred value.
3. Click **Save** to store configuration settings.

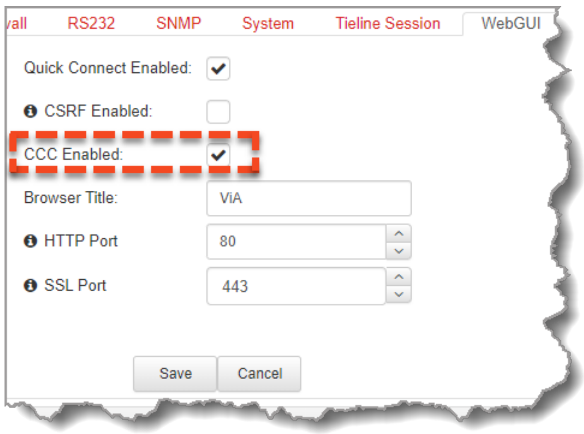


Important Note: Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

19.4 Enabling the Cloud Codec Controller

For 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.

19.5 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. 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	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.
3	Enable TieLink and Connection Interfaces	Enable TieLink and enable interfaces for use with TieLink (‘TieLink panel' in Toolbox, or TieLink screen in the codec).
4	Adjust TieLink Interface Priority	Adjust TieLink interface settings in the Toolbox web-GUI ‘TieLink panel’, e.g. priority of use.
5	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.
6	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. 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**.

1. Enable TieLink

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

2. Enable Interfaces

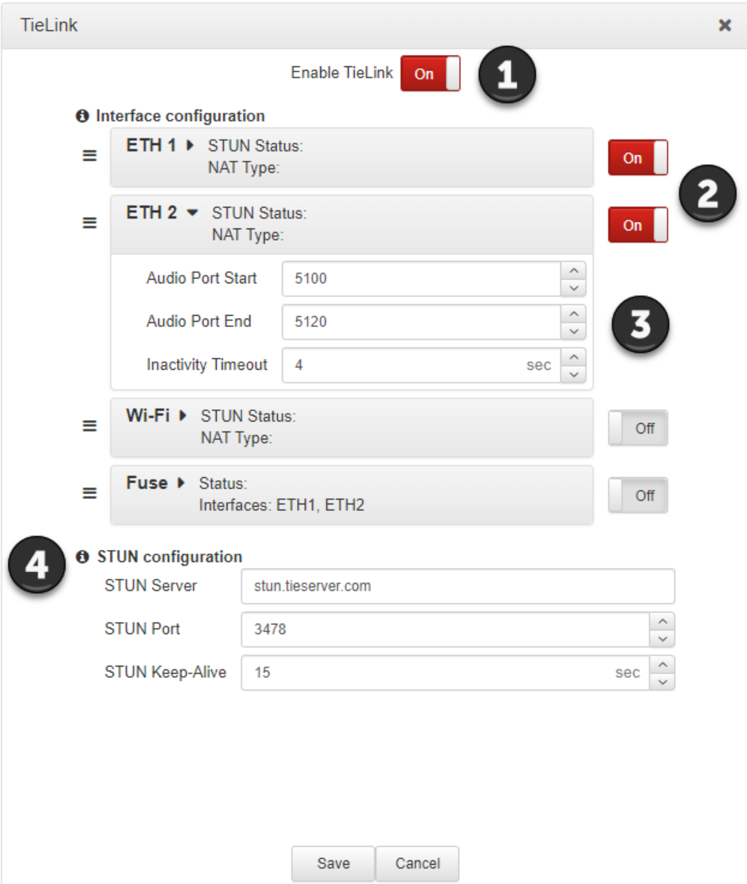
Click the **ON/OFF** button to enable each individual interface and/or select the interfaces used for **Fuse-IP** connections (Note: This is a simpler way to configure Fuse-IP. TieLink supports one connection to a single codec using Fuse-IP).

3. Adjust Tieline Interface Settings

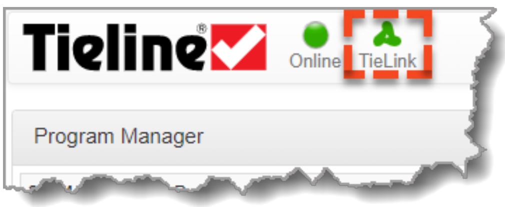
Adjust TieLink interface ports, inactivity timeouts and Fuse-IP interface selections.

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.



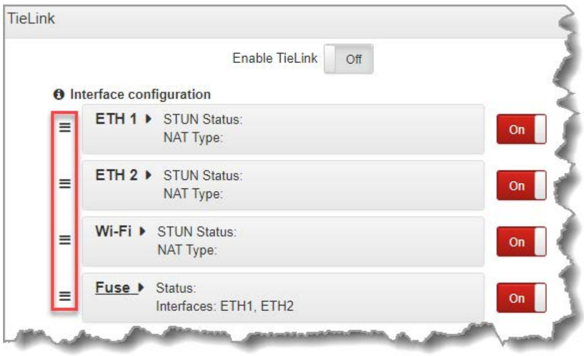
A green **TieLink Symbol** is displayed at the top of the Toolbox HTML5 Web-GUI when TieLink has been enabled successfully. Note: the symbol is orange if STUN is not successfully enabled on any interface.



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.

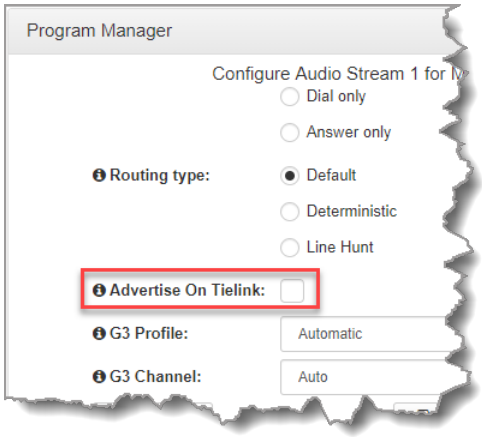
Configuring TieLink Connections in the Program Manager Panel

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.

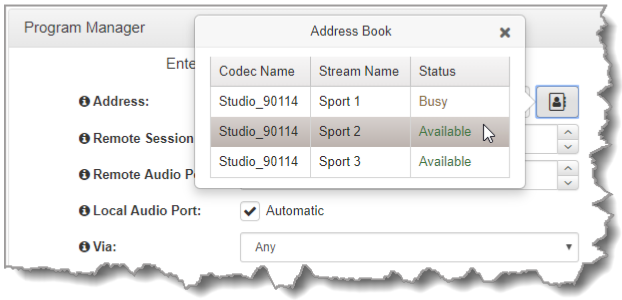
When creating a program select the **Advertise On TieLink** check-box to register answering audio streams on a loaded program with the TieLink Traversal server. These audio streams will be visible in TieLink contact lists which include this codec's audio streams. In other words, the answering audio streams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



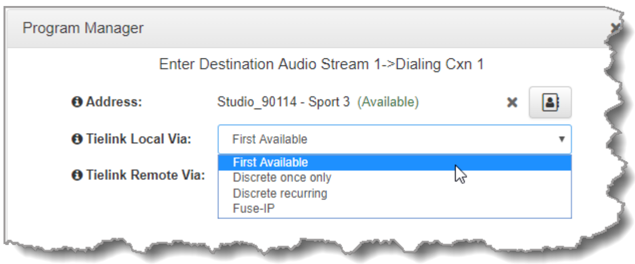
Select a Contact to Dial using TieLink

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**.

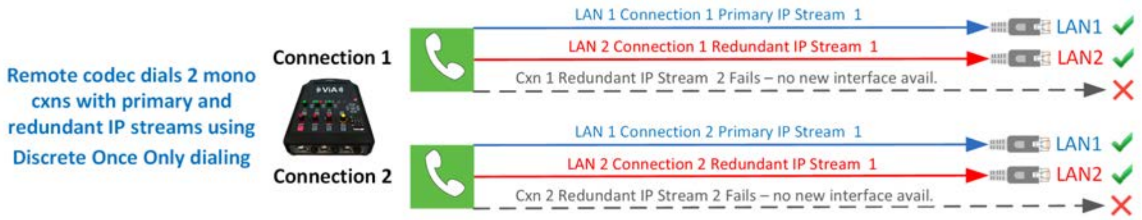


2. Select the interface options to use on local and remote codecs when dialing TieLink connections.

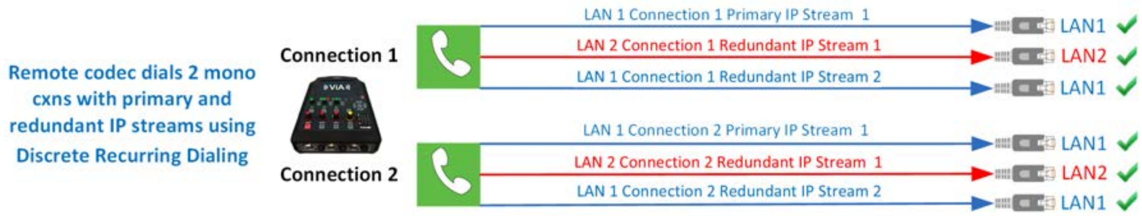


i. **First Available**: The first available interface that can successfully connect will be used to dial. If you dial two connections the same interface will be used for both connections, e.g. LAN1, even if another interface is available.

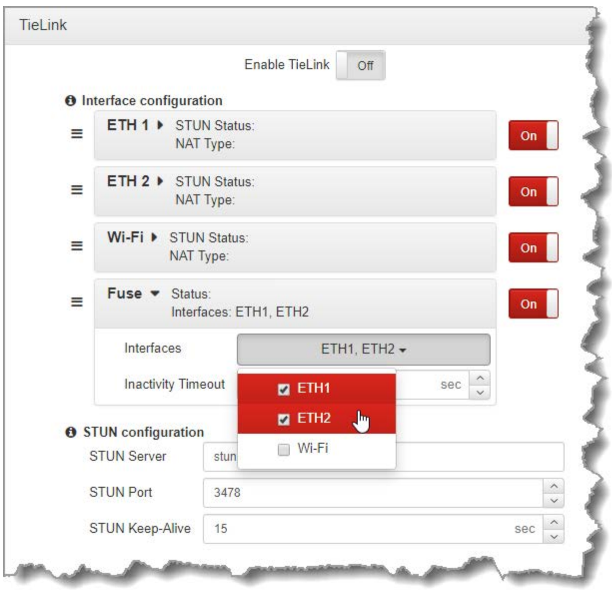
ii. **Discrete once only**: The first interface not being used by another connection will be used to dial. In the following example using two LAN interfaces, the codec connects a primary and redundant IP stream using different interfaces for each connection. A second redundant IP stream configured on 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 data bonding. Multiple interfaces can be selected and bonded to aggregate data and create a higher bandwidth connection. This is configured in the **TieLink panel** as displayed in the following example. 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:

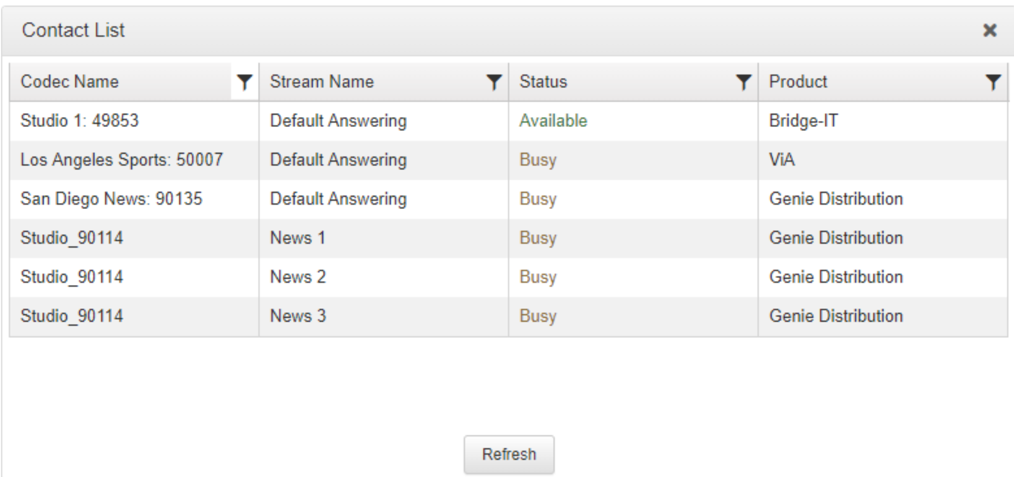
1. Based on the priority order of interfaces in the TieLink panel, a SmartStream PLUS connection may automatically use TieLink Fuse-IP as one of the connections if Fuse-IP is enabled at the dialing or answering codec. If TieLink Fuse-IP is chosen, then that connection will only succeed if the Fuse-IP 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.


Contact List

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

Refresh

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 **LCD 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.

19.6 Configure Fuse-IP Bonding

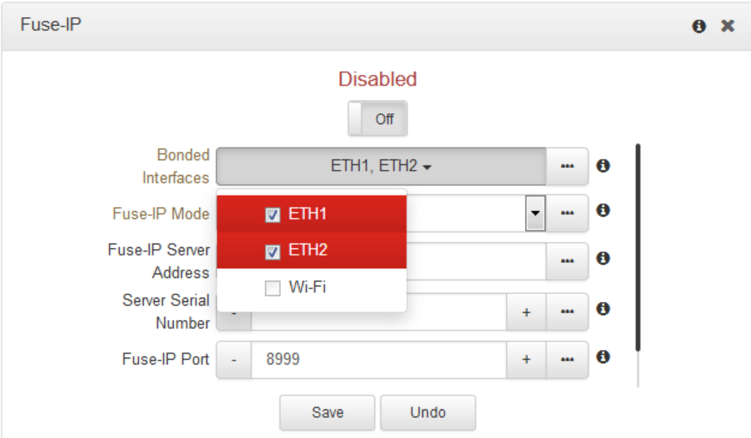
Tieline's proprietary Fuse-IP data aggregation technology uses a point-to-point tunnel between two codecs to bond multiple IP interfaces (peers). Fuse-IP automatically distributes data over any two bonded interfaces, which may include:

- Dual Ethernet LAN ports.
- An Ethernet LAN port with Wi-Fi.

See Connecting with Fuse-IP for more details on configuration using the codec front panel controls.

Configuring a Fuse-IP Server at the Studio

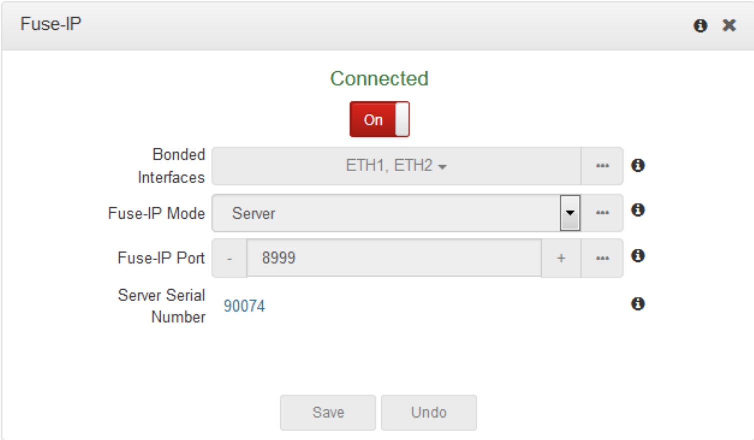
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Fuse-IP**.
2. Click the **Bonded Interfaces** drop-down menu to select the interfaces to be bonded.



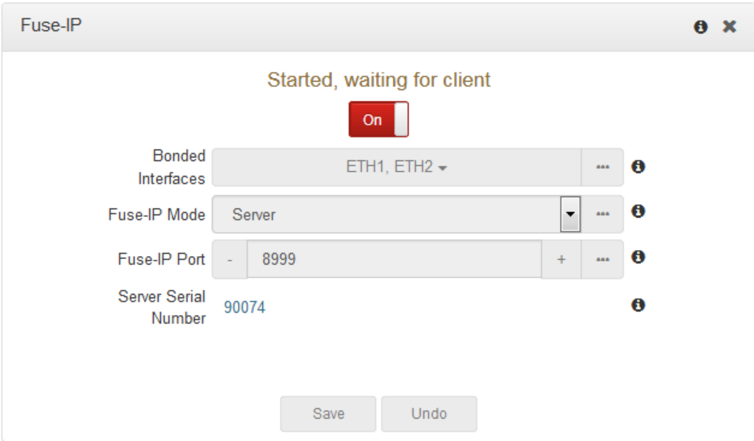
3. Click the **Fuse-IP Mode** drop-down menu and select **Server** if the codec is not initiating the connection.

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 **On/Off** button 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.

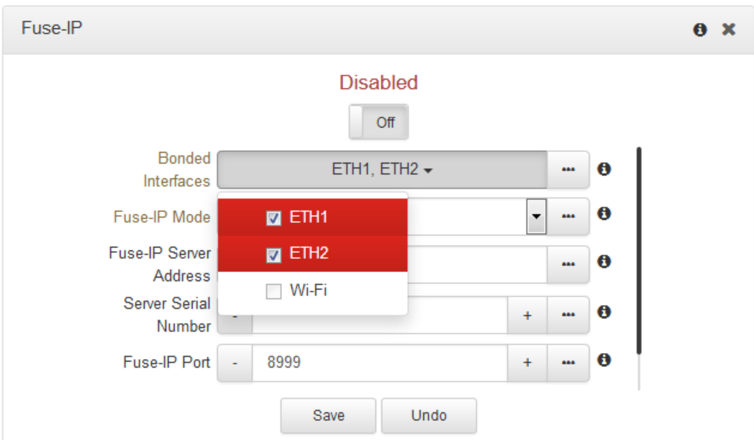


7. Double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after turning Fuse-IP **On**.

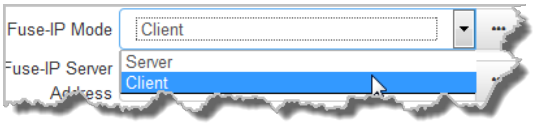


Configuring a Fuse-IP Remote Client

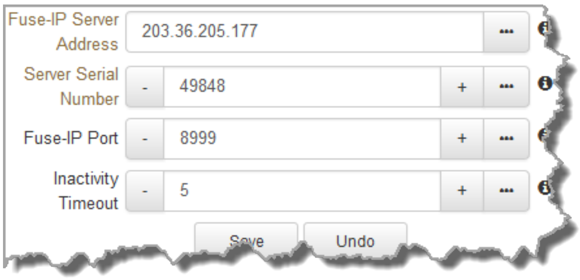
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Fuse-IP**.
2. Click the **Bonded Interfaces** drop-down menu to select the interfaces to be bonded.



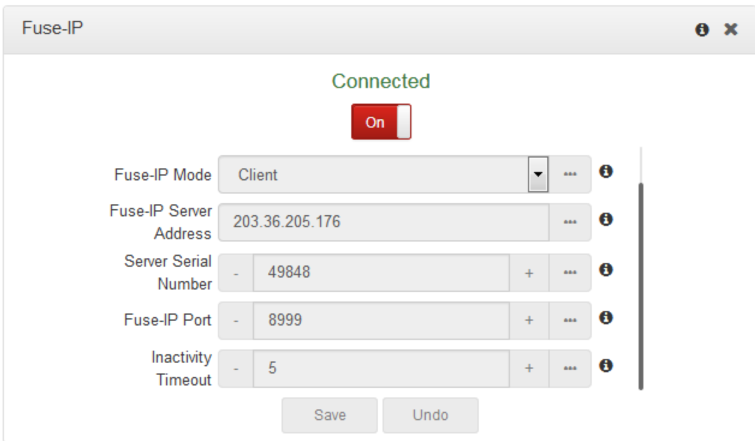
3. Click the **Fuse-IP Mode** drop-down menu to select **Client** as the Fuse-IP mode if dialing the studio codec. Note: the studio codec should be configured in server mode.



4. 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.



5. Click the **On/Off** button for Fuse-IP 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.

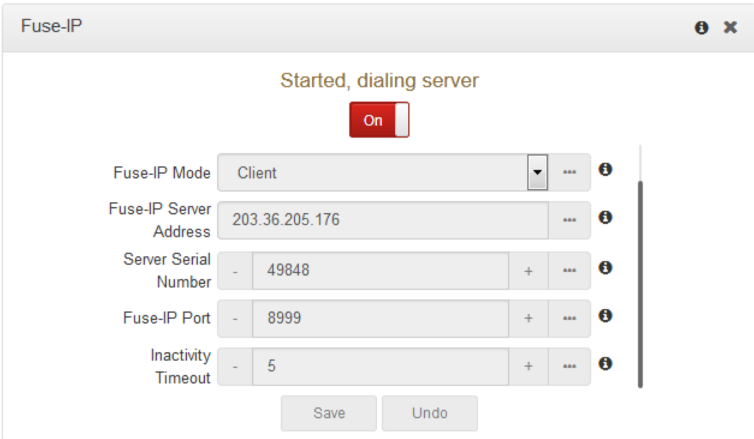


6. The status indicator is orange when Fuse-IP is enabled but no tunnel is created. It turns green when a tunnel is active.

Fuse-IP Enabled & no tunnel created

Fuse-IP enabled & tunnel created

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



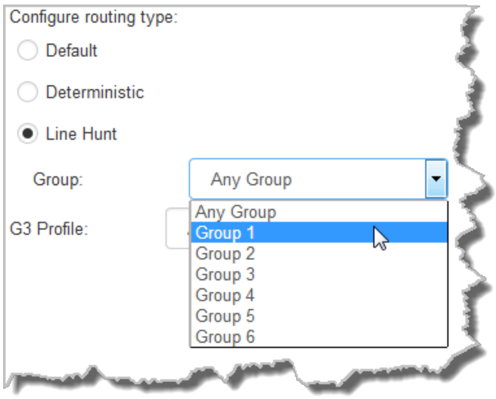
Important Notes:

- 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.

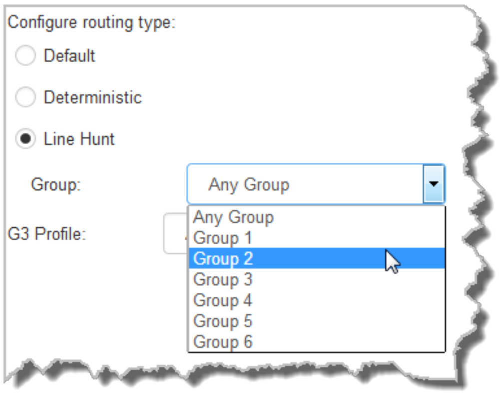
19.7 Line Hunt Call Answering

The codec supports line hunt call answering, whereby you can 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.

As an example, when creating a program which supports connecting six mono audio streams, select **Line Hunt** as the routing type. Then select **Group 1** for the first three audio streams, to route outgoing and incoming calls via inputs and outputs 1 to 3. These physical inputs and outputs on the codec can be routed to a particular studio or station.



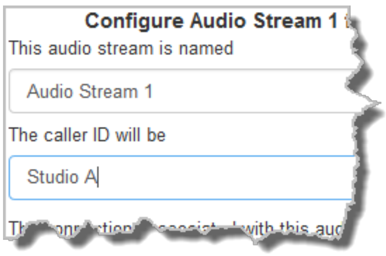
For the next three audio streams, select **Group 2** to route outgoing and incoming calls via inputs and outputs 4 to 6. These physical inputs and outputs on the codec can be routed to a different studio or station.



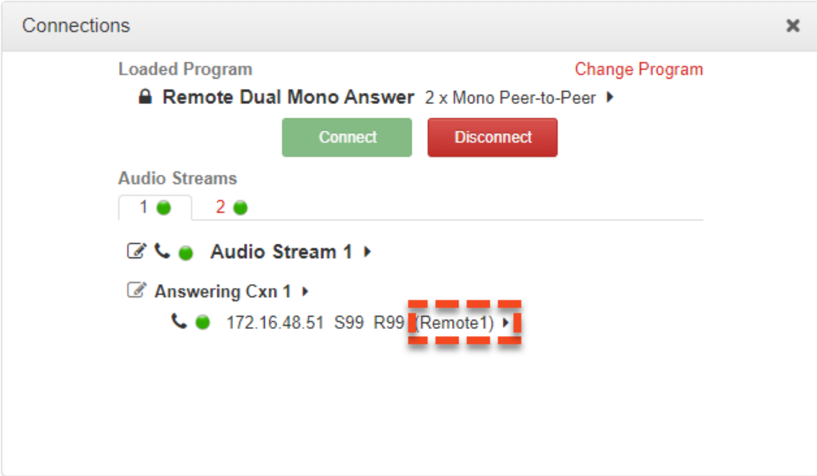
In a 6 stream answering program example, when **Any Group** is selected for all streams in **Line Hunt** mode, the codec will route all incoming calls on a first come, first served basis. The dialer doesn't have to specify a line hunt group, however they do need to select **Line Hunt** mode in the dialing program for this to function correctly.

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 codec dialing can display a designated **Caller ID** in the **Connection panel**. In the following example, **Remote1** has called into the codec using a specific caller ID, which is displayed next to the **Send** and **Return** link quality connection.



19.8 Configuring ISDN

Two slots are available for inserting optional ISDN modules into the codec. These can be configured using the codec front panel or the HTML5 Toolbox Web-GUI. See About ISDN Modules for additional information on ISDN.

You can use the HTML5 Toolbox Web-GUI to configure a dial and/or answer program with ISDN settings. You may also need to:

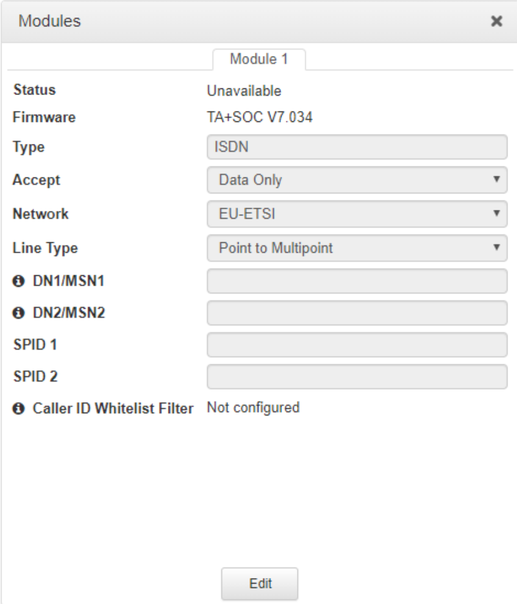
1. Configure ISDN module settings.
2. Configure ISDN Answering settings.

Important Note: For detailed information about connecting with other brands of codec using ISDN visit www.tieline.com

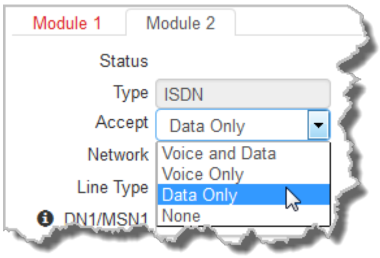
19.8.1 Configuring ISDN Modules

ISDN settings in the **Modules panel** determine how each codec module operates at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each ISDN module should connect. ISDN module settings may need to be adjusted depending on your country and network requirements.

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Modules** to view and configure ISDN site settings.

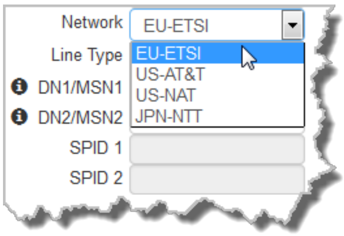


2. Click to select **Module 1** or **Module 2**.
3. Click the **Edit** button to configure settings.
4. Click the drop-down arrow for **Accept** to select whether to allow or disallow circuit switched voice and data calls. The default setting allows **Data only**.

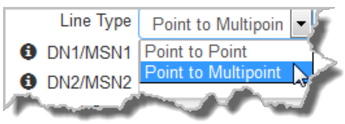


Important Note: G.711 is the algorithm used when **Voice Only** is selected.

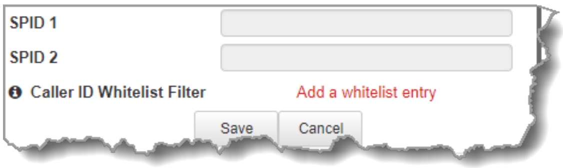
5. Click the drop-down **Network** arrow and select the **Network Type** corresponding to the region in which you are using the codec (see ISDN Module Settings for more details).



6. Click the drop-down **Line Type** arrow and select your preferred option. Ask your Telco whether your ISDN line is Point-to-Point or Point-to-Multipoint. By default select **Point-to-Multipoint**, unless your switch type is an AT&T 5ESS custom point-to-point.



7. If you are in the US enter DN and SPID numbers as required, or in other regions enter DN or MSN numbers as required.
8. Click **Add a whitelist entry** to only accept calls from numbers in the **Caller ID Whitelist Filter**. All calls are accepted if there are no whitelist entries.



9. Click **Save** when configuration is complete.

Important Notes:

Directory Numbers and Multiple Subscriber Numbers
Directory Numbers (DN) in North America and Multiple Subscriber Numbers (MSN) in the rest of the world are simply phone numbers associated with an ISDN B channel, like lines listed in a typical phone directory. Your Telco will normally supply 2 DN/MSN numbers for each pair of B channels. However, these numbers may or may not be associated with a specific B channel.

Often broadcasters prefer to predict which B channel will answer an incoming call to ensure audio routing is consistent. However, if a DN or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number. Programming DN/MSN numbers for each B channel allows the codec to ignore calls without matching DN/MSN numbers. This is the best way to answer calls from codecs in a predictable manner.

SPID Numbers in North America
ISDN relies on an initialization procedure for associating Service Profiles with specific terminating equipment (e.g. your audio codec) rather than lines. In the US Telcos assign a Service Profile ID (SPID) number which assists in identifying different ISDN services across the network. Your Telco must provide a SPID for each B channel you order when connecting over US-Nat or US-AT&T networks in the US. A SPID is not required when using the AT&T PTP protocol.

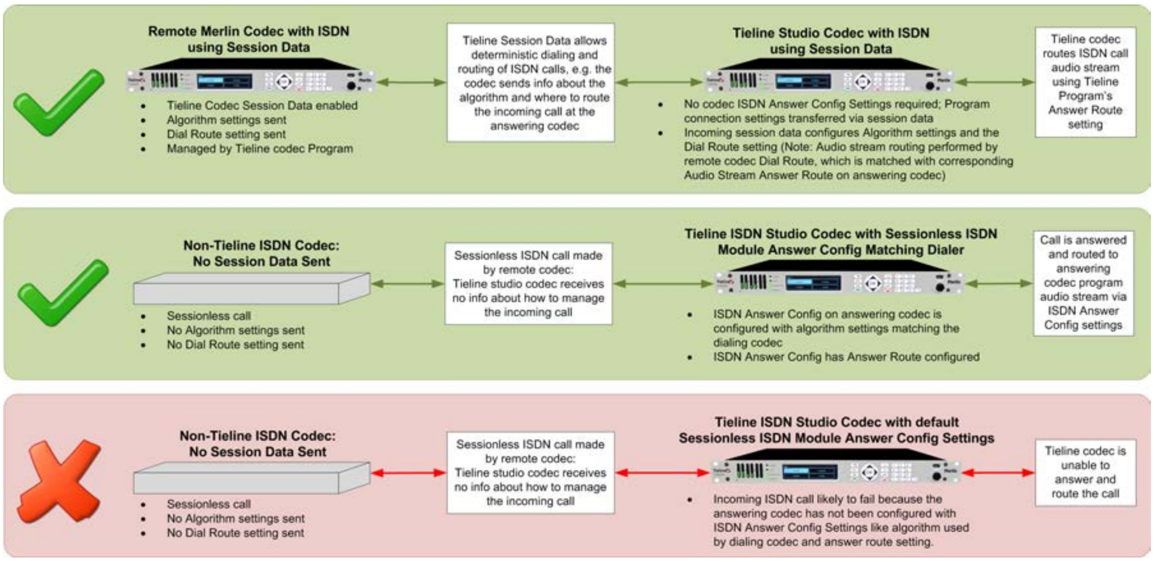
Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch. The switch then associates the SPID with a specific Service Profile and directory number.

Note: SPID numbers normally include the phone number and additional prefix or suffix digits up to 20 digits long.

19.8.2 Configuring ISDN Answering

ISDN Answer Configs are used to determine how codec ISDN modules will behave when answering ISDN calls.

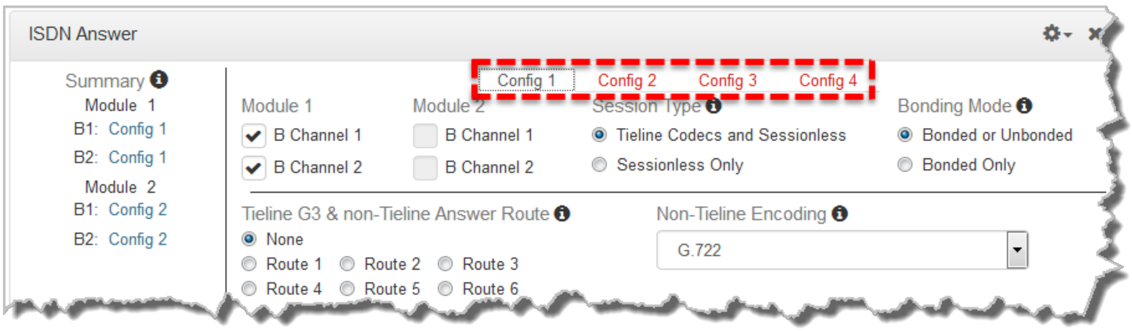
The following image explains the difference between answering calls from Tieline codecs sending session data, and non-Tieline codecs making sessionless ISDN calls. Codecs sending Tieline Session Data contain all the information required to connect, e.g. algorithm and audio stream routing settings. When answering sessionless calls it is necessary to configure the answering codec with an **ISDN Answer Config**, which tells the answering codec how a sessionless call will try and connect.



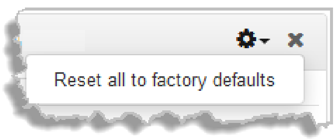
It is possible to save up to four different **ISDN Answer Configs**, which allow up to 4 ISDN B channels to be individually configured for unique answering behaviors. ISDN answering can be configured to suit:

- Hardware available in the codec, i.e. the number of B channels available.
- Expected dialing behaviors, e.g. if B channels should bond or not, and whether audio streams need to use **Dial** and **Answer Route** tags.
- The type of call being received by the codec, e.g. Tieline (with Tieline Session Data) versus non-Tieline sessionless calls.
- The algorithm expected when receiving sessionless calls.

Each of the four available **Configs** allows you to select which B channel or channels are used to answer a call or calls from incoming ISDN codecs. Up to 4 B channels can be selected if 2 ISDN modules are installed in the codec.



To reset ISDN answering to default settings click the **Options symbol** in the top right-hand corner of the panel and select **Reset all to factory defaults**.

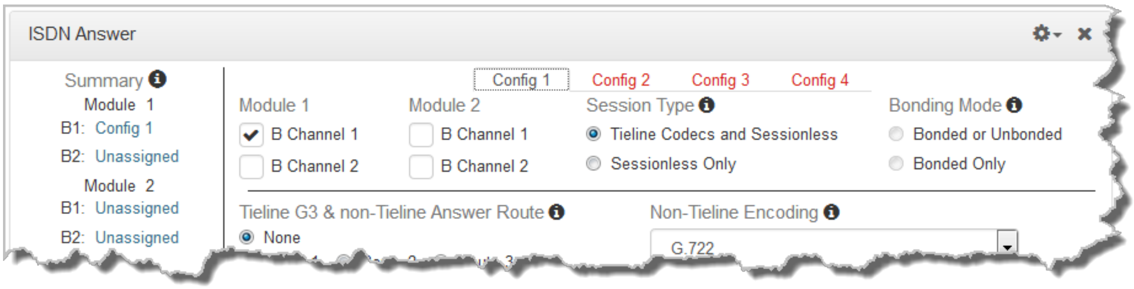


Important Note: B channels can only be selected once and a **Summary** of allocated B channels is displayed on the left-hand side of the **ISDN Answer** panel.

Single B Channel Config

To use a single 64kbps B channel for a connection (e.g. a 1 x Mono Peer-to-Peer audio stream):

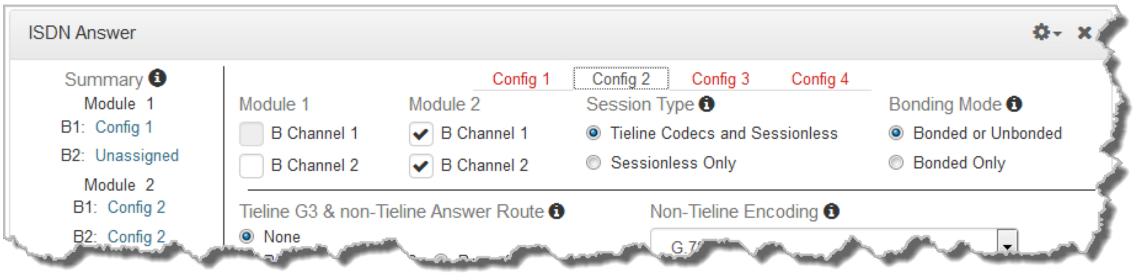
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **ISDN Answer**.
2. Click to select a **Config**.
3. Select a B channel from those available and then click **Save**. The connection is not bonded if only one B channel is selected.



Multiple B Channel Bonding Config

A point-to-point audio stream can also bond multiple B channels to create higher bandwidth connections.

1. Click to select a **Config**.
2. Select multiple B channels in the **Config**. Note: In the following example, two B channels from **Module 2** have been selected within **Config 2**. Note that **B Channel 1** in **Module 1** has already been assigned in **Config 1** and is therefore greyed out and unavailable in **Config 2**.



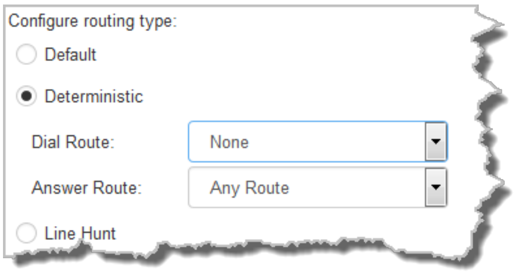
3. Configure the bonding setting that best suits the audio stream associated with this **Config**. **Bonded or Unbonded** is the best setting in most situations.

Bonding Setting	Behavior
Bonded or Unbonded (May Bond)	Calls using the same algorithm from the same Tieline codec, or sessionless calls, will attempt to bond when received. Calls using incompatible algorithms will not be bonded
Bonded Only	Will only bond compatible algorithms. This mode will reject incompatible calls which cannot be bonded, e.g. G.711 and G.722

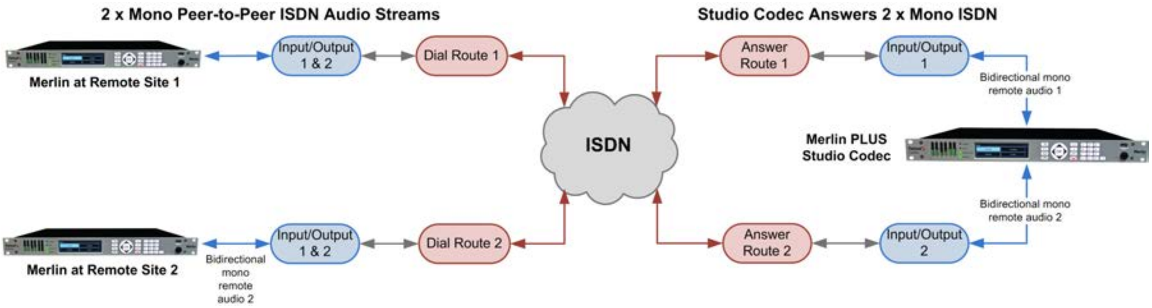
4. Click **Save** to apply changes to the **Config**.

Dial and Answer Route Settings in Programs

Dial Route and **Answer Route** tags allow you to associate a B channel (or channels) in a **Config** with a particular incoming audio stream from either Tieline G3 or non-Tieline codecs. This is not necessary in simple point-to-point ISDN audio stream configurations, however it is very useful in multiple audio stream codecs using multiple B channels. When dialing Tieline to Tieline over ISDN using the Merlin or Genie family of codecs, you can configure a **Dial Route** in the dialing codec's program and a corresponding **Answer Route** in the answering codec's program. This will ensure a particular audio stream is routed between two codecs consistently. This feature is not available in Tieline G3 codecs, so an **Answer Route** should be used for deterministic routing when receiving calls from these codecs.



In principle, the concept of 'routes' operates similarly to how audio ports are used to route multiple audio streams over IP. Selecting different IP audio port numbers allows users to define which incoming IP audio stream is routed to a specific answering audio stream configuration on the codec. This ensures inbound calls from multiple codecs can be consistently routed to the same answering codec audio streams, and therefore the same inputs and outputs. Following is an example of how to consistently route incoming ISDN audio streams using dial and answer routes.

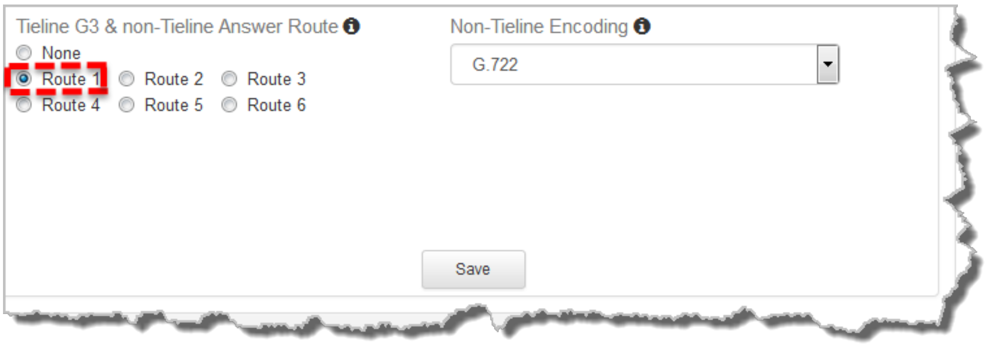


Answer Routes for Non-Tieline (Sessionless) or Tieline G3 ISDN Calls

In some situations you may receive a call from a non-Tieline codec which doesn't support session data and **Dial Route** tags. In this situation you can still specify the audio stream **Route** on the answering codec using **Config 1-4** in **ISDN Answer**. You can also select the default algorithm.

For example, if a call from a non-Tieline codec is being received via **B Channel 1** on **Module 1** (i.e. no **Dial Route** has been specified in the dialing codec):

1. Click to select a **Config**.
2. Select a **Route** for this B channel in one of the four **Configs** within the **ISDN Answer** panel, e.g. **Route1**, then select the default **Non-Tieline Encoding** algorithm to use when answering calls from non-Tieline codecs (default setting is **G.722**).



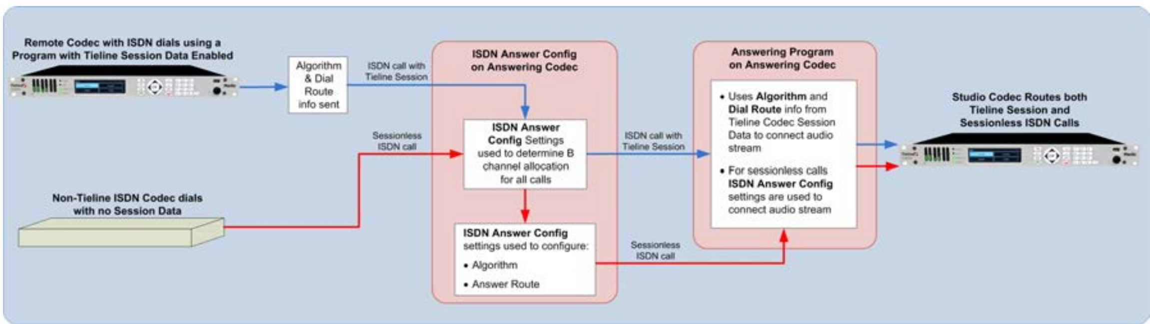
3. Click **Save** when configuration is complete to store the new **Config** settings.
4. This configuration will associate an incoming call to this B channel with a corresponding **Answer Route** configured in the answering program, e.g. **Answer Route 1** in the following image.

More detailed information about how to configure the codec to answer and route multiple sessionless ISDN calls is available in Using ISDN Answer Routes for Sessionless ISDN Calls. This uses examples to explain how to set up consistent deterministic routing of multiple incoming sessionless calls.

Answering both Tieline Session and Sessionless ISDN Calls

Leave the **Sessionless Only** check-box in the **ISDN Answering Config** unchecked if the codec is expected to receive ISDN calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, when the codec answers a call it initially expects to receive Tieline session data from the dialing codec and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (i.e. a non-Tieline codec is calling, or a Tieline codec with session data disabled), it will use the settings in the **ISDN Answering Config** instead.

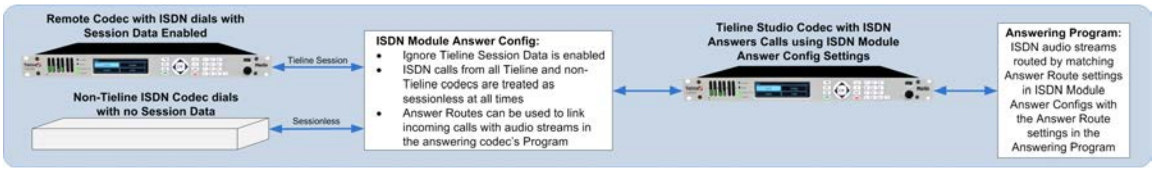
The following image displays how the answering codec will behave in this mode when receiving calls from both Tieline and non-Tieline codecs.



Allow Answering of Sessionless ISDN Calls Only

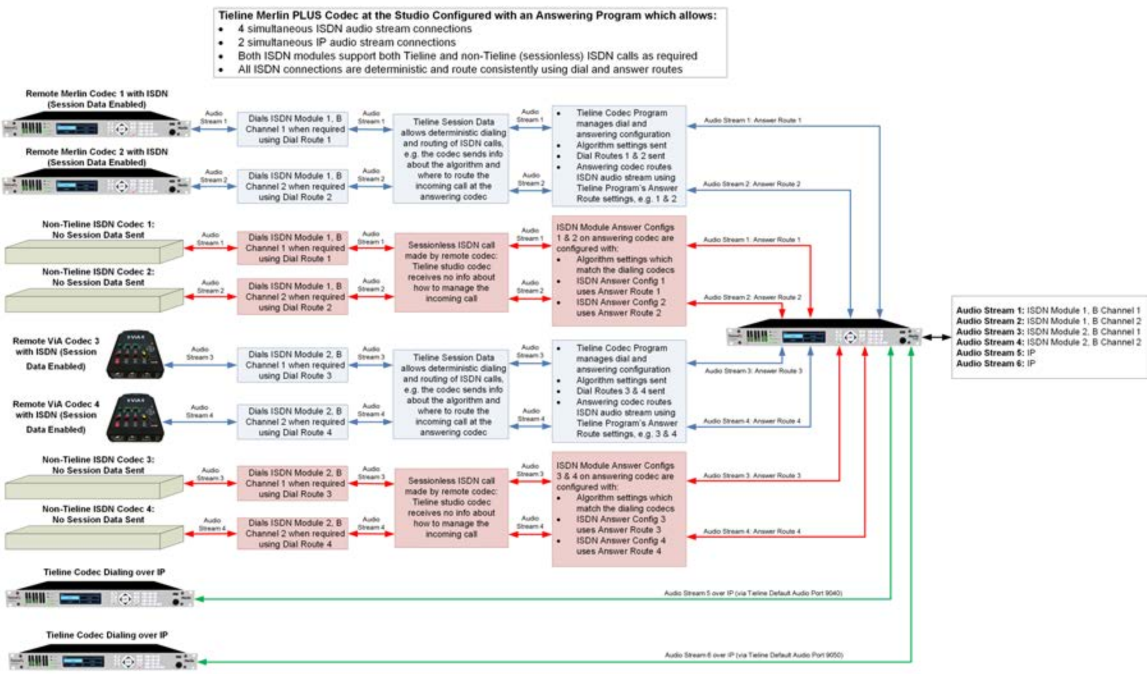
Select **Sessionless Only** when answering ISDN calls from non-Tieline codecs only. When **Sessionless Only** is selected, the codec will not wait to receive the Tieline session data. This reduces the time taken to answer an inbound sessionless call.

The following image displays how the answering codec will respond with **Sessionless Only** selected, i.e. calls from both Tieline and non-Tieline codecs are always regarded as sessionless.



Answering Multiple ISDN Calls from Tieline and non-Tieline Codecs

Tieline codecs capable of answering multiple incoming audio streams can be configured to answer both Tieline session data and sessionless ISDN calls at different times. They can also support connections using other transports such as IP or POTS. The following example shows how a Tieline codec can be configured to answer up to 4 separate mono ISDN calls at different times from both Tieline and non-Tieline codecs, as well as two mono IP audio streams.



Default Answering Settings

When a B channel is not associated with a **Config** it inherits the following default settings:

- Tieline Session
- Unbonded
- G.722 algorithm
- Audio route: None

19.9 Configuring POTS

Two slots are available for inserting optional POTS modules into the codec. These can be configured using the codec front panel or the HTML5 Toolbox Web-GUI. See About POTS Modules for additional information on POTS.

You can use the HTML5 Toolbox Web-GUI to configure a dial and/or answer program with POTS settings. You may also need to:

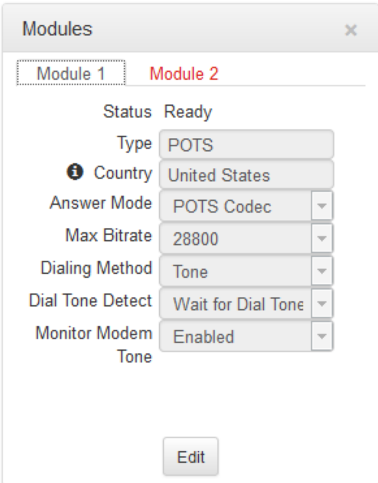
1. Configure POTS module settings.
2. Configure POTS Answering settings.

19.9.1 Configuring POTS Modules

POTS settings in the **Modules panel** menu determine how your codec will connect at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each module should connect. The default **Config** settings for POTS modules are designed to suit Tieline codecs. These settings will need to be adjusted to connect to non-Tieline POTS codecs or connect in **Analog Phone** mode.

Configuring POTS G5 Modules

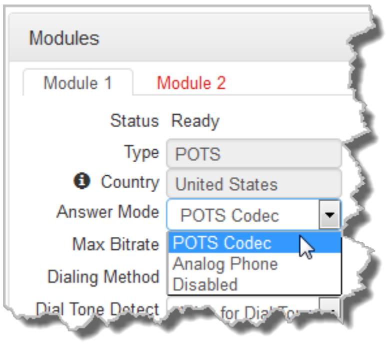
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Modules** to view and configure POTS site settings.



Important Notes: The POTS module **Status** is displayed in the **Modules panel**. **No Phone Line** is displayed when a cable is detached; **Ready** is displayed when a cable is attached and the line voltage is good.

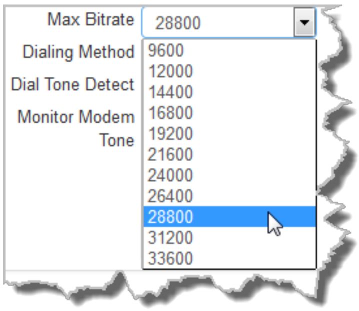
2. Click to select **Module 1** or **Module 2**.
3. Click the **Edit** button to configure settings.
4. Click the drop down arrow to adjust the **Answer Mode** and select how the module in the codec will be able to answer incoming POTS calls. Options include:

- **POTS Codec:** allows the POTS G5 module to receive incoming audio data over a POTS line.
- **Analog Phone:** configures the POTS G5 module to receive a standard analog phone call.
- **Disabled:** disables the POTS G5 module from receiving a **POTS Codec** or **Analog Phone** call.



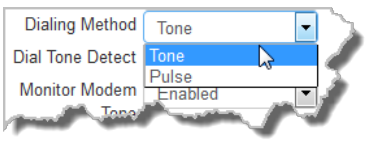
Calls are answered based on the **POTS Answer settings** in **Config 1** & **2**. Adjustments to these **Config** settings are not normally necessary when connecting between Tieline codecs. They are usually adjusted when connecting to non-Tieline codecs over POTS (see Configuring POTS Answering for more info).

5. Click the **Max Bitrate** drop-down arrow to adjust the maximum bit rate (dialing and answering). The default setting is **28800** (28.8kbps) and this only affects **POTS Codec** calls. The range of the setting is 9.6kbps to 33.6kbps. Even if the line is capable of establishing a connection at a higher bit rate, the **Max Bitrate** setting is the highest bit rate that will be attempted. Reducing this value can improve connection reliability on poor quality lines. If two codecs are not configured with the same setting, they will attempt to connect at the lowest of the two **Max Bit rate** settings.



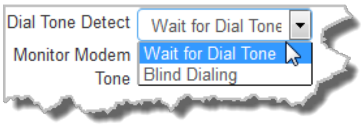
Important Note: G5 POTS modems initially attempt to establish a link at the lowest **Max Bitrate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

6. Click the drop-down arrow for **Dialing Method** to select **Tone** (DTMF) or **Pulse** dialing over POTS Codec connections. Tone dialing is used always when the **Answer Mode** is **Analog Phone**.

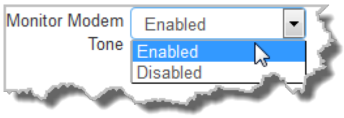


7. Click the drop-down arrow for **Dial Tone Detect** to select either:

- **Wait for Dial Tone**: The module will only be allowed to dial when a dial tone is present on the line.
- **Blind Dialing**: Allows the module to dial when no dial tone is present.



8. Click the drop-down arrow for **Monitor Modem Tone** to select either **Enabled** or **Disabled**.



When enabled the module will allow audio monitoring of modem tones during connection in **POTS Codec** mode via the phone input. By default, the following phone input monitoring rules apply when multiple POTS G5 modules are installed in a codec and multiple POTS connections are dialed.

Module 1	Module 2	Audio Rule
POTS Codec (Monitor Modem Tone)	POTS Codec (Monitor Modem Tone)	The phone input receives a mix of modem tone audio from both modules
POTS Codec (Monitor Modem Tone)	Analog Phone	The phone input receives analog phone input audio only and mutes modem tone monitoring
Analog Phone	Analog Phone	The phone input receives audio from the oldest active connection only

Important Notes:
- Modem tone monitoring will work even if **Phone Input Enable** is **Off** via **Settings > Audio > Phone Input > Phone Input Enable [Off]**.
- Modem tone monitoring is only enabled during the initial connection training and negotiation period in **POTS Codec** mode.
- The monitoring volume can be adjusted using the codec front panel via **Settings > Audio > Phone Input > Level**, or by opening the **Inputs panel** in the Web-GUI and adjusting the **Phone** input volume slider.

9. **Country** displays the current country setting in the codec. To adjust this setting select **Settings > System > Country**.
10. Click **Save** when configuration is complete.

19.9.2 Configuring POTS Answering

It is possible to store a different **POTS Answer Config** for each POTS module installed in the codec. POTS answering can be configured to suit:

- The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline (Sessionless).
- Expected dialing behaviors and encoding, e.g. whether audio streams use **Route** tags and which algorithm is used.

If you answer a call from a non-Tieline codec you will need to create an answering "Config" to determine which module in the codec will answer the call and the settings used when connecting.

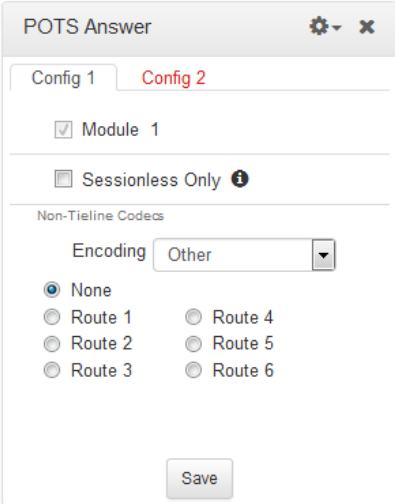
Important Notes:
- **POTS Answer Config** settings are applied to **POTS Codec** connections and not **Analog Phone** connections.
- When receiving a call from a Tieline codec with session data enabled (i.e. not **Sessionless**), the algorithm setting from the dialing codec overrides the setting in the **POTS Answer Config** menu.

POTS Config Settings

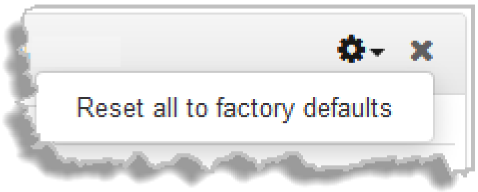
The default **POTS Answer** module **Config** settings, which can be viewed in the **POTS Answer panel** are:

- **Tieline Codecs** Session Data.
- The **Other** algorithm.

This configuration will accept the settings from an incoming Tieline codec when it dials with session data enabled. It will also allow the codec to answer a call from a Comrex POTS codec supporting the **Other** algorithm setting.

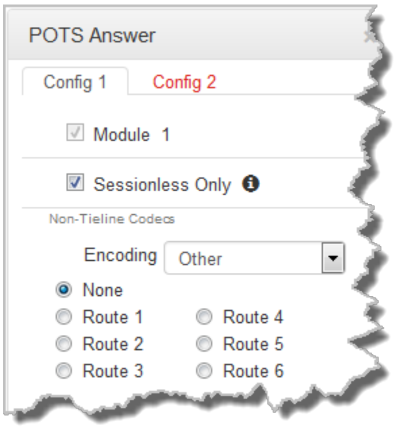


To reset POTS answering to default settings click the **Options symbol** in the top right-hand corner of the panel and select **Reset all to factory defaults**.



Answering Calls from Non-Tieline POTS Codecs

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **POTS Answer**.
2. Click the **Edit** button to configure settings.
3. Select the **Sessionless Only** check-box when only non-Tieline codecs are dialing a Tieline codec over POTS. This allows you to select the default encoding setting and **Route** the incoming call to a nominated audio stream via a corresponding **Answer Route** in the answering codec program if required.

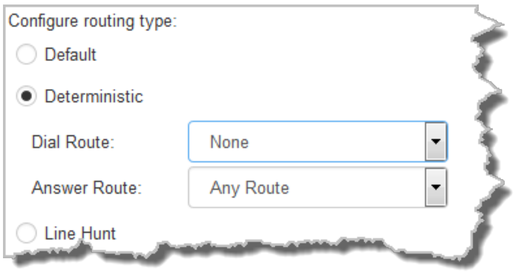


4. Click **Save** to apply changes to the **Config**.

> **Important Note:** Select **Other** in the **Encoding** drop-down menu when connecting to Comrex® Vector, Matrix® and BlueBox® codecs. On the Comrex codec select its "Music" algorithm. Please note that 9.6kbps connections are not supported by the Comrex codecs.

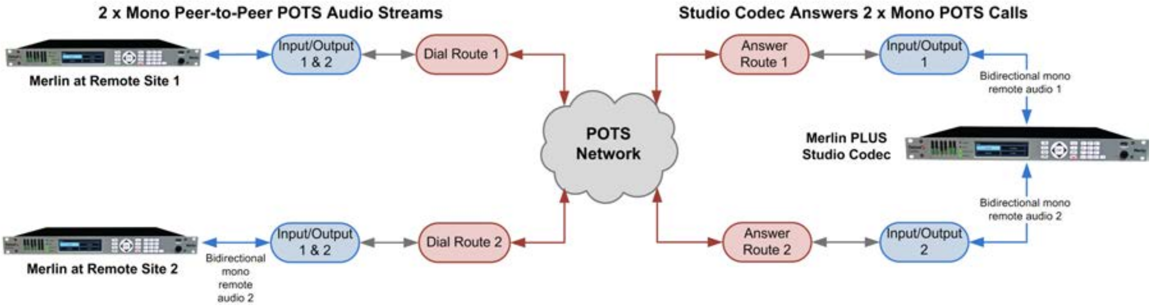
Dial and Answer Route Settings in Programs

Dial Route and **Answer Route** tags allow you to associate a POTS **Config** with a particular incoming audio stream from either Tieline or non-Tieline codecs.



In principle, this operates similarly to how audio ports are used to route multiple audio streams over IP. Selecting different IP audio port numbers allows users to define which incoming IP audio stream is routed to a specific answering audio stream configuration on the codec. This ensures inbound calls from multiple codecs can be consistently routed to the same answering audio streams, and therefore the same inputs and outputs.

This is not necessary in simple point-to-point POTS audio stream configurations, however it is very useful in multiple audio stream codecs which support POTS connections. When dialing Tieline to Tieline over POTS using the Merlin or Genie family of codecs, you can configure a **Dial Route** in the dialing codec's program and a corresponding **Answer Route** in the answering codec's program. This will ensure a particular audio stream is routed between two codecs consistently.

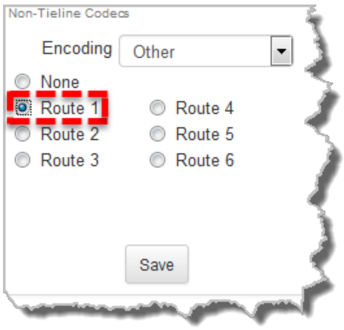


Answer Routes for Non-Tieline POTS Codecs

In some situations you may receive a call from a non-Tieline POTS codec which doesn't support **Dial Route** tags. In this situation you can still specify the audio stream **Route** on the answering codec using **Config 1 or 2** in **POTS Answer**. You can also select the default algorithm.

For example, if a call from a non-Tieline codec is received via POTS **Module 1** (i.e. no **Dial Route** has been specified in the dialing codec):

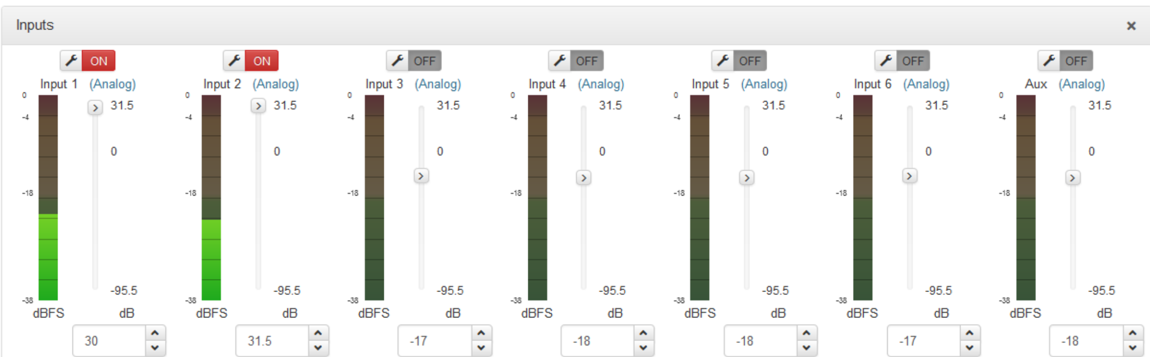
1. Click the **Edit** button to configure settings.
2. Select an answering **Route** for this POTS module in one of the two **Configs** available in the **POTS Answer panel**, e.g. **Route1**, then select the default **Encoding** algorithm **Other** (Note: **Other** is used for connecting to Comrex POTS codecs).



3. Click **Save** to store the new **Config** settings.
4. This will associate the incoming call with a corresponding **Answer Route** configured in the codec answering program, e.g. **Answer Route 1** as displayed in the following example.

19.10 Configuring Input/Output 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 the Auxiliary input; this is disabled by default.

Adjusting Audio Levels

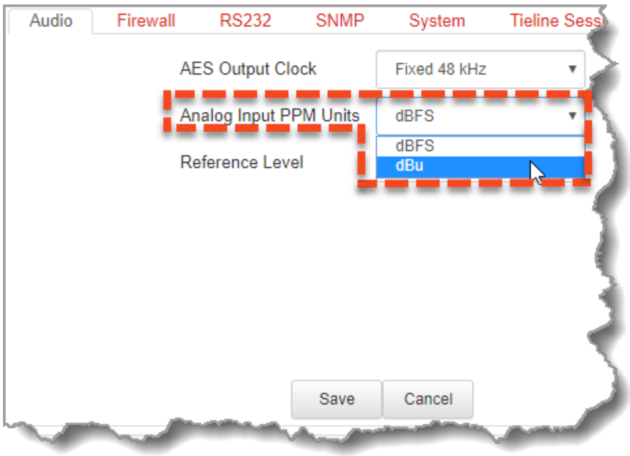
To adjust input audio levels, click on the input slider and drag it to the desired input gain level. Alternatively, click the arrows below a PPM meter to incrementally increase or decrease the input level in 0.5dB steps. Input levels on the **Input panel** should be set to ensure audio peaks average at the first yellow indications on the PPM meters, which represents nominal 0 VU at -18dBFS. Audio levels should also be verified using the meters in the **PPMs panel**.

Important Notes: 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 **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **Audio** and then click **Edit**.
3. Click the **Analog Inputs PPM Units** drop-down menu and select **dBU**, then click **Save**.

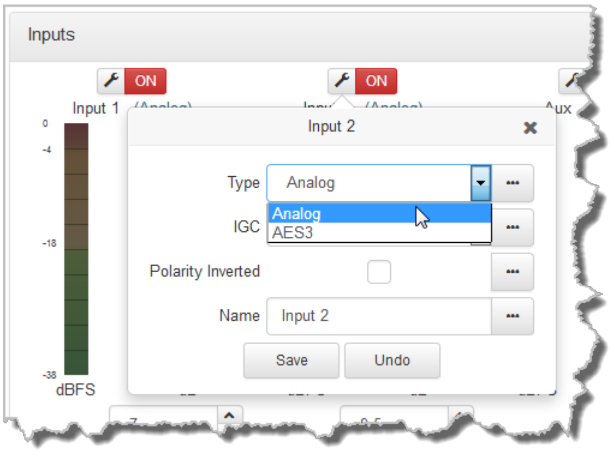


Configuring Input Settings

Selecting Analog and Digital Audio Sources

Codec inputs are configured for analog line level audio sources by default.

1. Click the **Input Settings** symbol.
2. Select **Type** and then **Analog** or **AES3**.

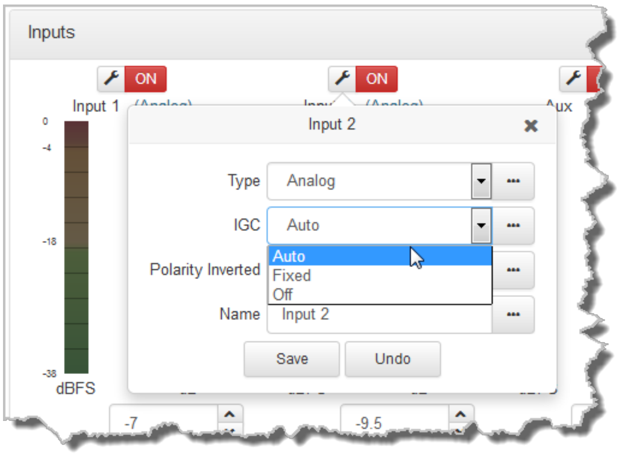


3. Click **Save** to confirm the new setting.

Important Notes: See Configuring AES3 Input Audio for more information about the 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.

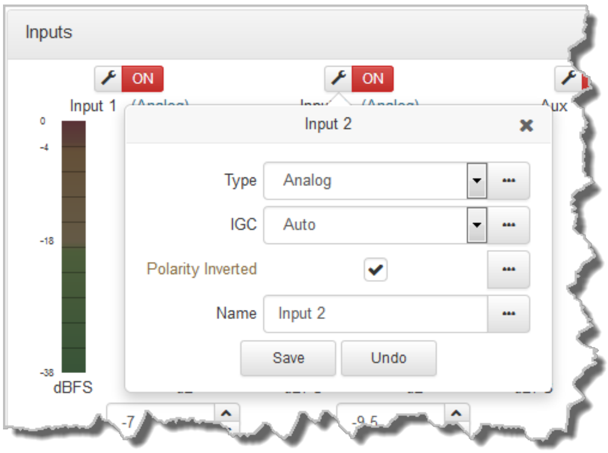


3. Click **Save** to confirm the new setting.

Invert Polarity

Select the **Polarity Inverted** check-box to reverse the polarity of an analog or digital input.

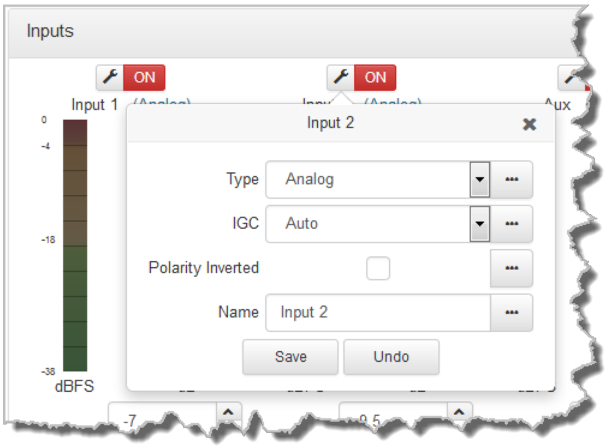
1. Click the **Input Settings** symbol on the input you want to adjust.
2. Click to select the **Polarity inverted** check-box.



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

Renaming Inputs

1. Select 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.



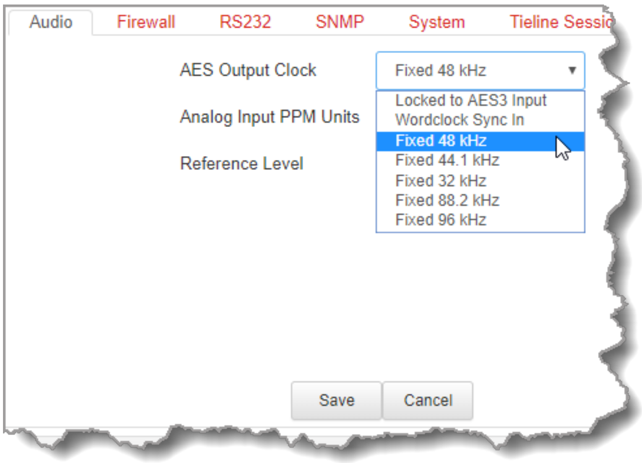
3. Click **Save** to confirm the name change.

Important Note: When the auxiliary input (**AUX IN**) is **On** the default mixer configuration sends audio to all inputs. If you are not using the auxiliary input ensure it is **Off** to avoid additional noise in program audio.

AES3 Output Sample Rate Configuration

The AES3 output sample rate can be configured using the HTML5 Toolbox Web-GUI.

1. Open the HTML5 Toolbox Web-GUI and click **Settings**, then click **Options** to open the **Options panel**.
2. Select **Audio** and then click **Edit**.
3. Click the **AES Output Clock** drop-down menu to select your preferred **AES Output Clock** setting, then click **Save**.



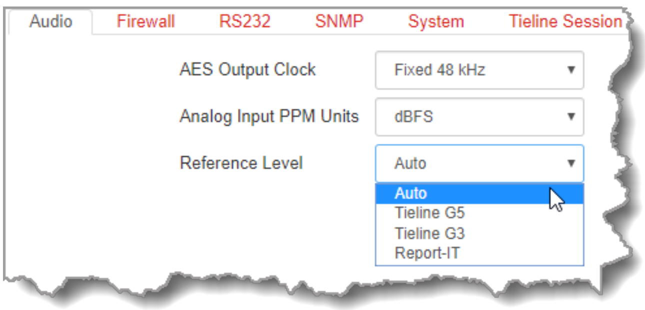
Audio Reference Levels

By default, the **PPM METERS** on the front of the codec, 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.

Important Note: 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.

To adjust this setting:

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



19.11 Configure Mono or Stereo Peer-to-Peer Programs in Merlin

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 Merlin Peer-to-Peer Programs: Dialing
2. Configuring a Merlin Backup Connection or Auto Reconnect
3. Configuring Merlin to Answer Connections

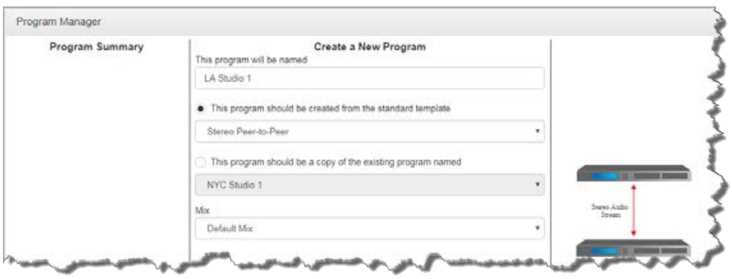
For more information about programs and audio streams within programs see the section titled About Program Dialing. Note: The following display how to configure a dial and answer program, with a backup connection. If you want the codec to either dial or answer only, select one of these options and the wizard will automatically display a subset of relevant screens to allow you to configure the codec correctly.

Configuring Merlin Peer-to-Peer Programs: Dialing

Important Notes: Before you start program configuration please note:

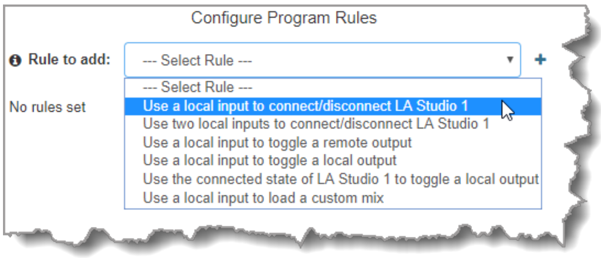
- You cannot edit a program when it is currently loaded in the codec.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- POTS is not supported for stereo audio stream connections.
- To learn more about programs see the section titled About Program Dialing.

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 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 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.

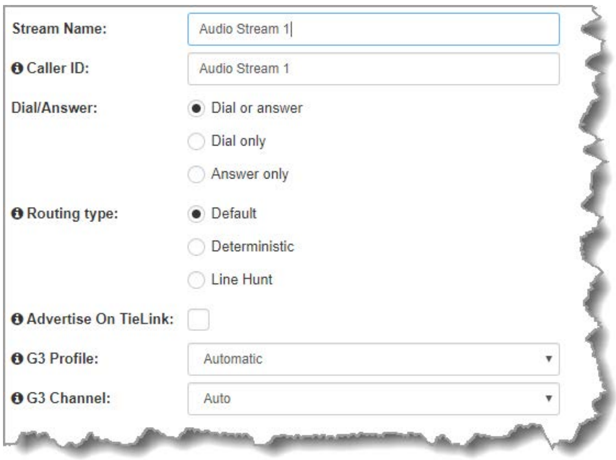
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Notes for Rules:

- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet 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). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be 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. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	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 ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	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.

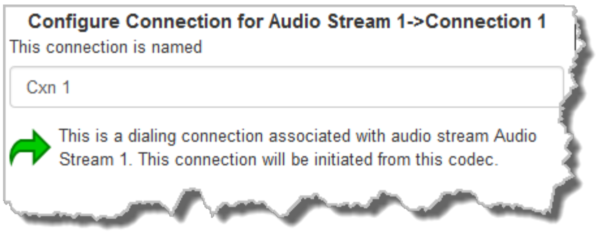
Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

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 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

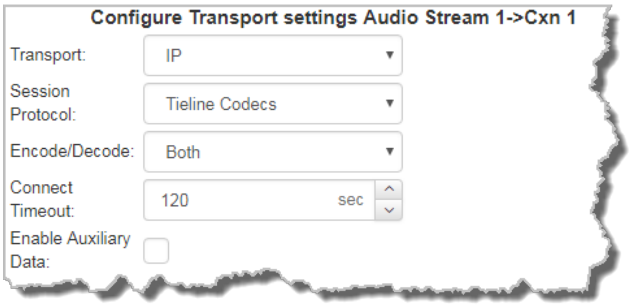
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

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.

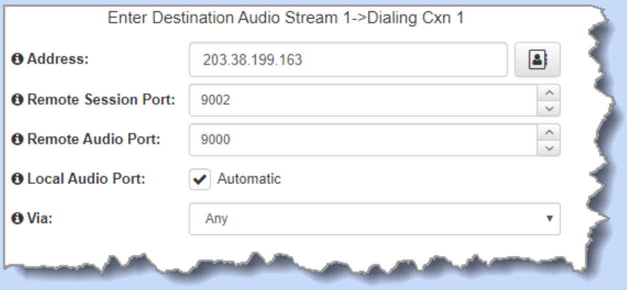


Important Note:
- If you select **Sessionless** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.
- See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure destination codec dialing and encoding settings:

IP

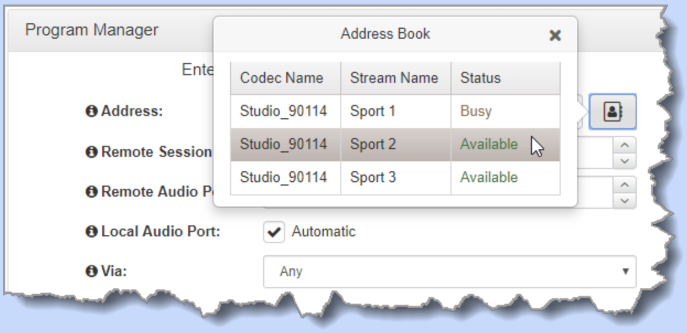
For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.



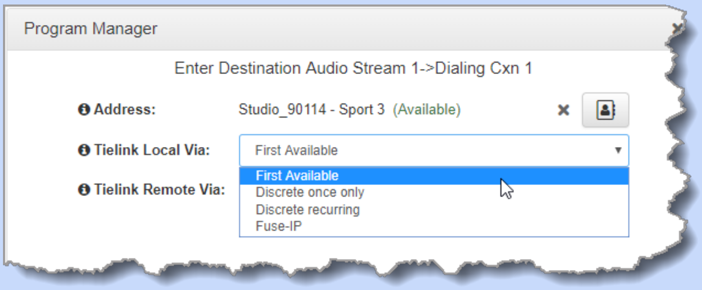
Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), 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. 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**.

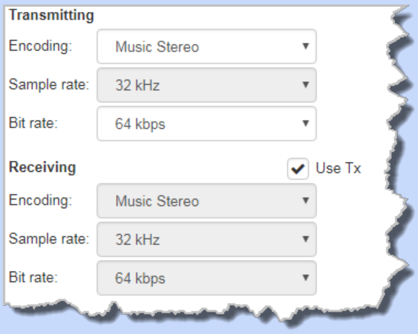


2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see [Configuring TieLink Settings] for more info.



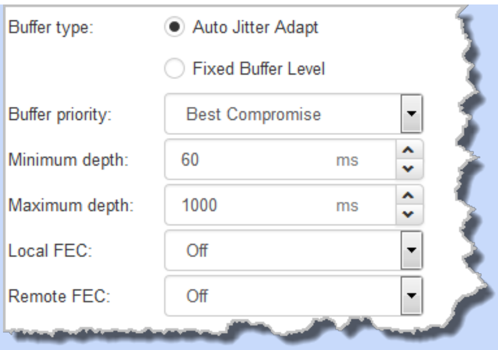
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a failover connection or SmartStream PLUS for this audio 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.



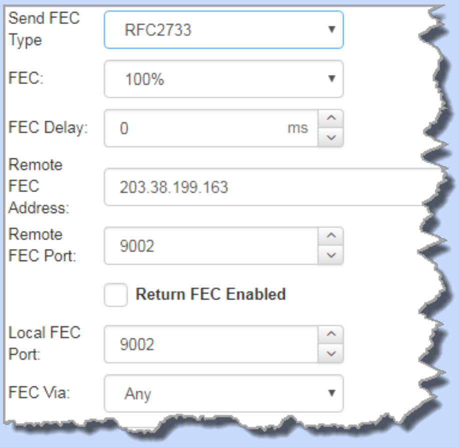
For IP connections click to configure:

- **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.



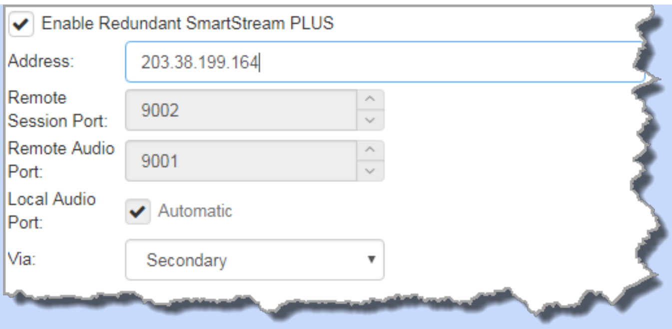
Important Notes:

- If you select **Sessionless** or **SIP** as the **Session Protocol** then RFC-compliant FEC is displayed. 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. 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 **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. 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.

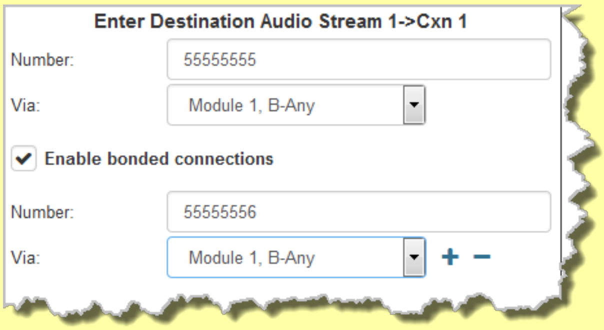


Important Notes:

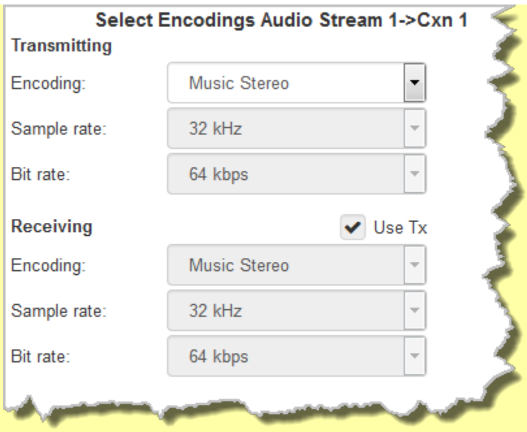
- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three 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/

ISDN

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.



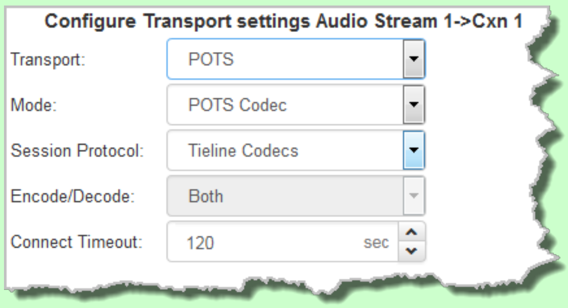
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).



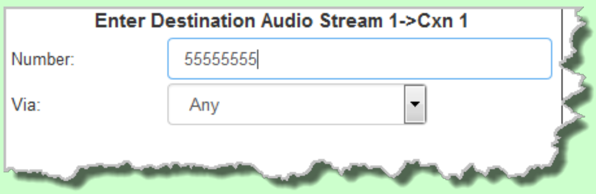
Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2 to** specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



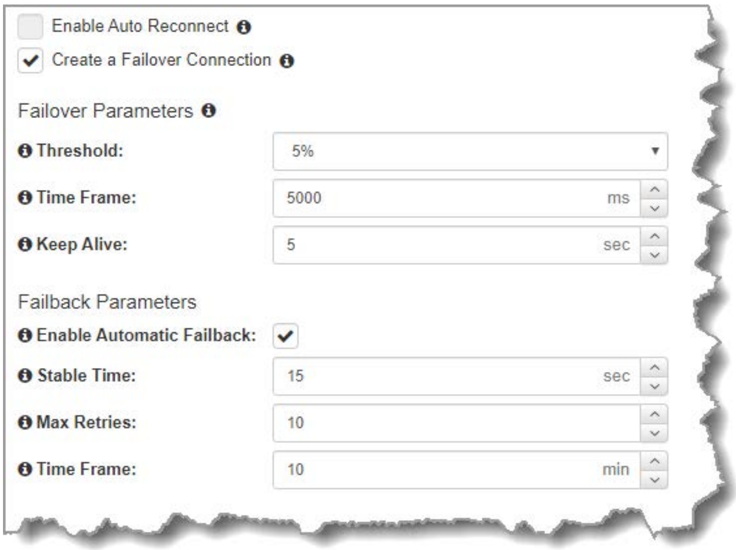
Dialing settings for this POTS audio stream are now complete.

Configuring a Merlin 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 you are configuring.

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.

1. Click to select the check-box for **Create a Failover Connection**. Adjust the 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**.

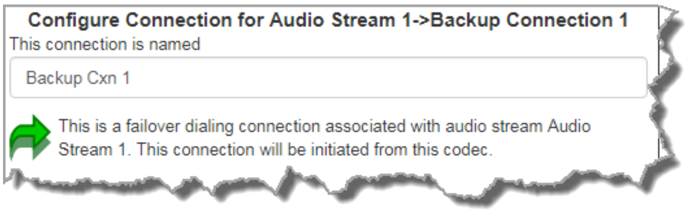
The codec will always attempt Failback if any of the following conditions are met:

- 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 within the following table can be used to assist with failover connection configuration.

	Screen Display	Description
1	Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame against which lost data is measured
3	Keep Alive	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	The time frame used to measure the number of fail back retries attempted

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

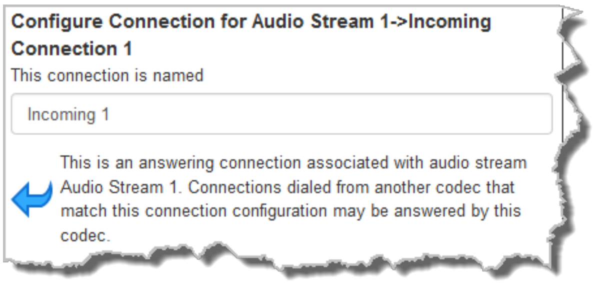


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you configured the primary connection.

Configuring Merlin to Answer Connections

The codec is capable of being configured to accept calls via different transports (e.g. IP and ISDN), or to accept calls using different audio ports. If you are configuring the codec to allow it to answer one or more incoming audio stream connections:

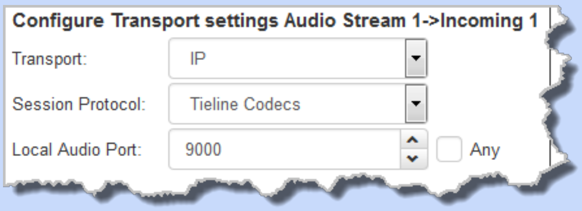
1. Enter a name for the answering connection and click **Next**.



2. Configure the transport settings:

IP

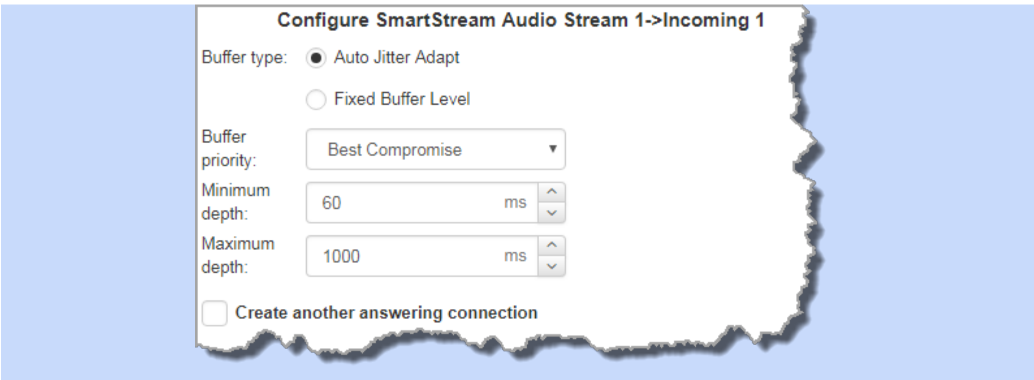
For IP select the **Session Protocol** and **Audio Port**.



Important Note: The **Local Audio Port** is the port 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 **Local Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream connection. Click to deselect the **Any** check-box to adjust this setting.

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, or
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.



ISDN For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

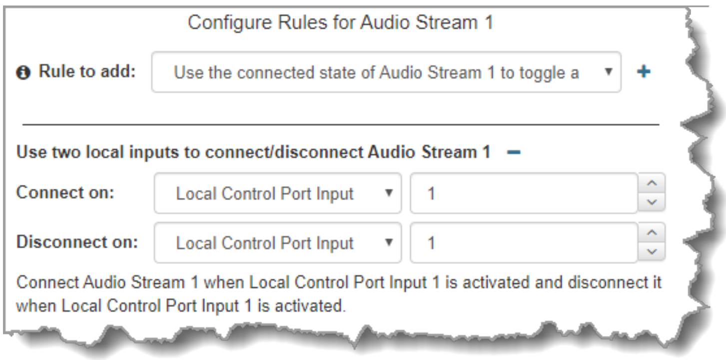
POTS For POTS, settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

3. 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 Program** to save the program at this point.
   iii. Click **Next** to configure rules options.

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.

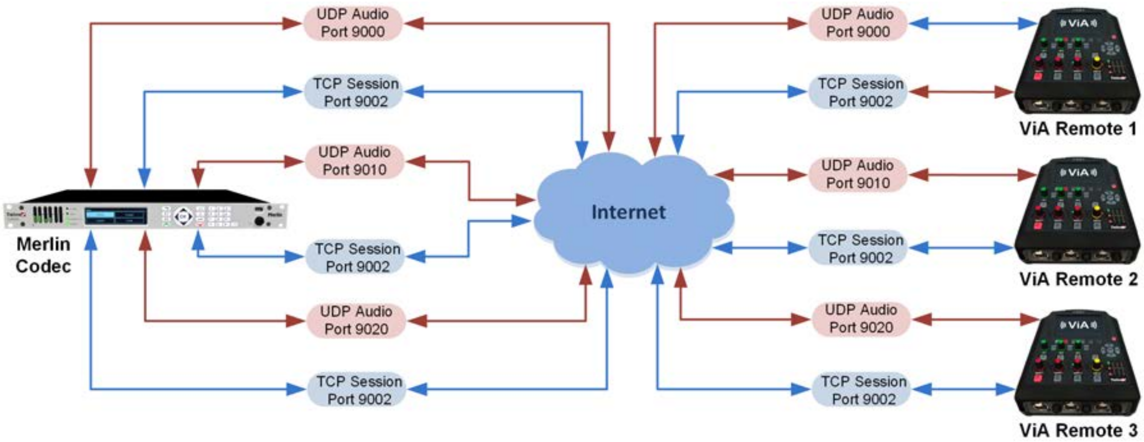


Important Note: Connection-related rules are not displayed in **Answer only** programs.

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**, **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.

19.12 Configure 2 or 3 Mono Peer-to-Peer Answering Programs in Merlin

Two or three simultaneous mono peer-to-peer audio stream connections can be configured.



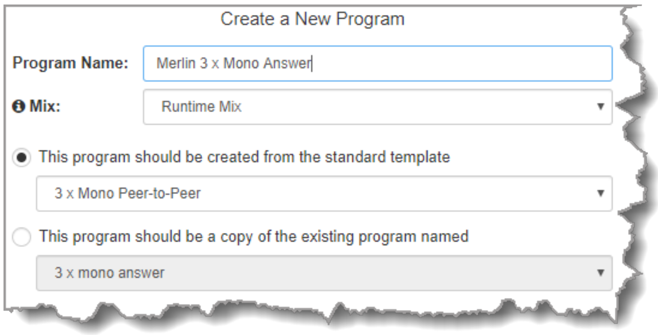
The following program wizard procedure displays the configuration screens to create an answering connection for each incoming call. See Configuring Merlin Point-to-Point Programs for more details about individual settings within the program wizard.

Routing 2 or 3 Incoming Mono Audio Streams to Specific Codec Outputs

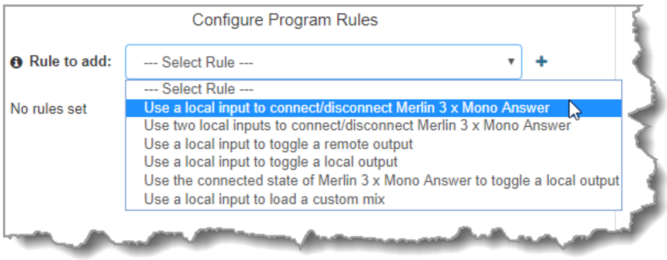
Important Notes: Before you start program configuration please note:

- You cannot edit a program when it is currently 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 type of program. 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 and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- To learn more about programs see the section titled About Program Dialing.

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 **2 x Mono Peer-to-Peer**, or **3 x Mono Peer-to-Peer** (the following example uses 3 x mono), or if you want to use an existing program as a template, select this option. Then click **Next**.



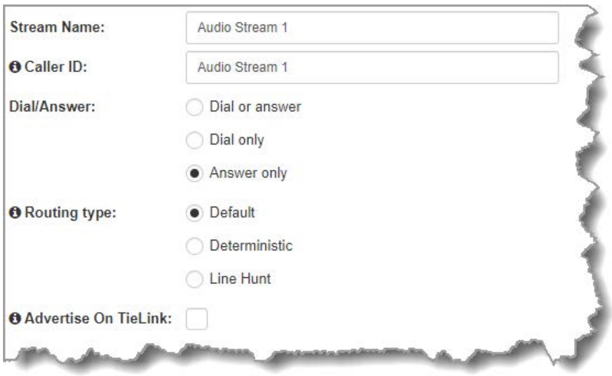
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



Important Notes for Rules:

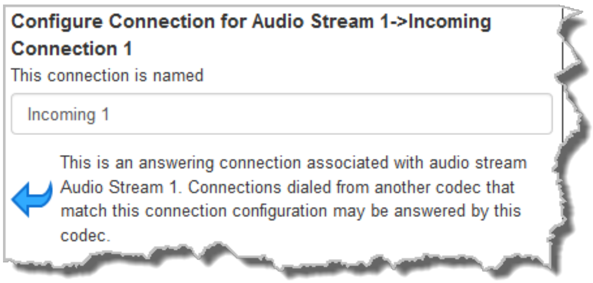
- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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**, add a **caller ID** and configure the codec to **Answer only**. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



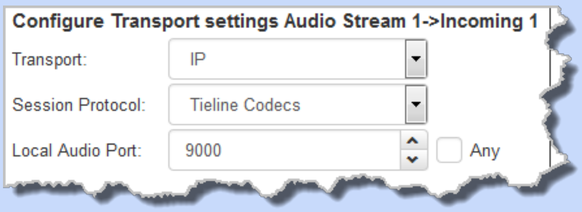
Routing Type Options:	
Default	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
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.

5. Enter the name of the connection in the text box, then click **Next**.



6. Configure the transport settings:

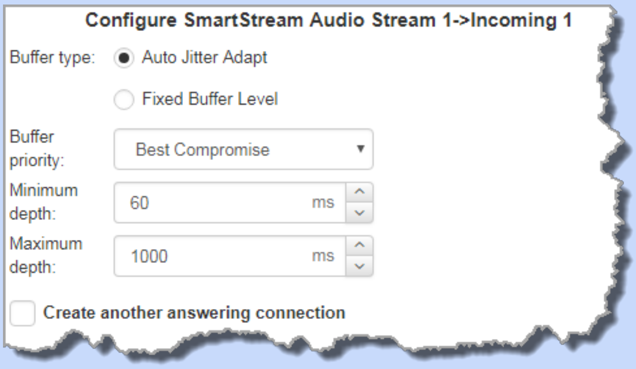
For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next** to specify individual algorithm, jitter buffer and FEC settings, or create another answering connection.



Important Note: 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 **Local Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream connection. Deselect the **Any** check-box to adjust this setting. A codec dialing this connection and using the port specified will always be routed to output 1 on the codec receiving the call.

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.



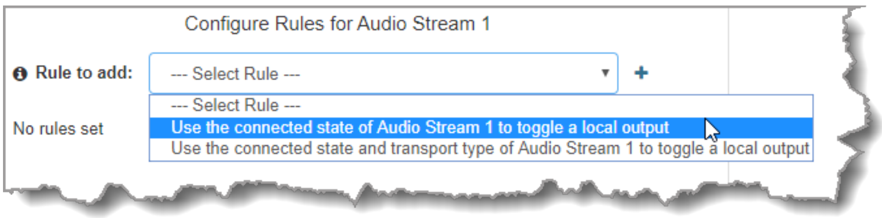
ISDN module answering settings are used if you select ISDN as the connection transport. For more details see Configuring ISDN Answering.

POTS module answering settings are used if you select POTS as the connection transport. For more details see Configuring POTS Answering.

7. After configuring all settings there are 3 options:

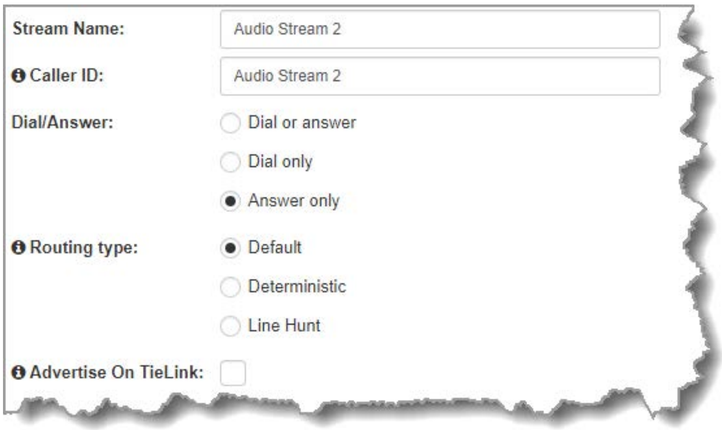
   i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
   ii. Click **Next Stream** to configure the second audio stream.
   iii. Click **Next** to configure rules options.

8. To configure new stream related 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.

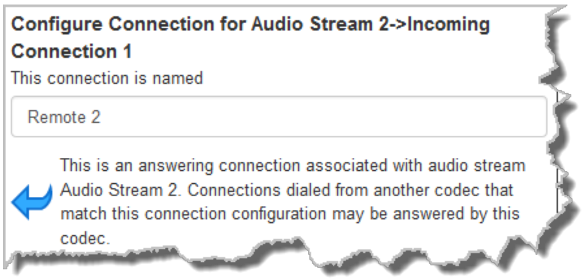


Important Notes: Connection-related rules are not displayed in **Answer only** audio streams.

9. Click **Next** and enter the next **Stream Name**, add a **caller ID** and configure the codec to **Answer only**. Then click **Next**.

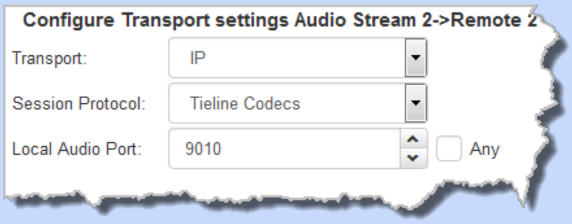


10. Enter the name of the second audio stream connection in the text box and click **Next**.



11. Configure the transport settings:

For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected.



Important Note: When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Local Audio Port** is automatically configured as UDP audio port 9010 by default for the second audio stream connection. Click to deselect the **Automatic** check-box to adjust this setting. A codec dialing this connection and using the port specified will always be routed to output 2 on the codec receiving the call.

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.

ISDN module answering settings are used if you select ISDN as the connection transport. For more details see Configuring ISDN Answering.

POTS module answering settings are used if you select POTS as the connection transport. For more details see Configuring POTS Answering.

12. Continue through the steps in the wizard to complete configuration in the same way as the first connection was configured. Note: Configure the third audio stream in the same way as the first two when creating a **3 x Mono Peer-to-Peer** program.
13. Click **Save Program** at the end of this process. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**.

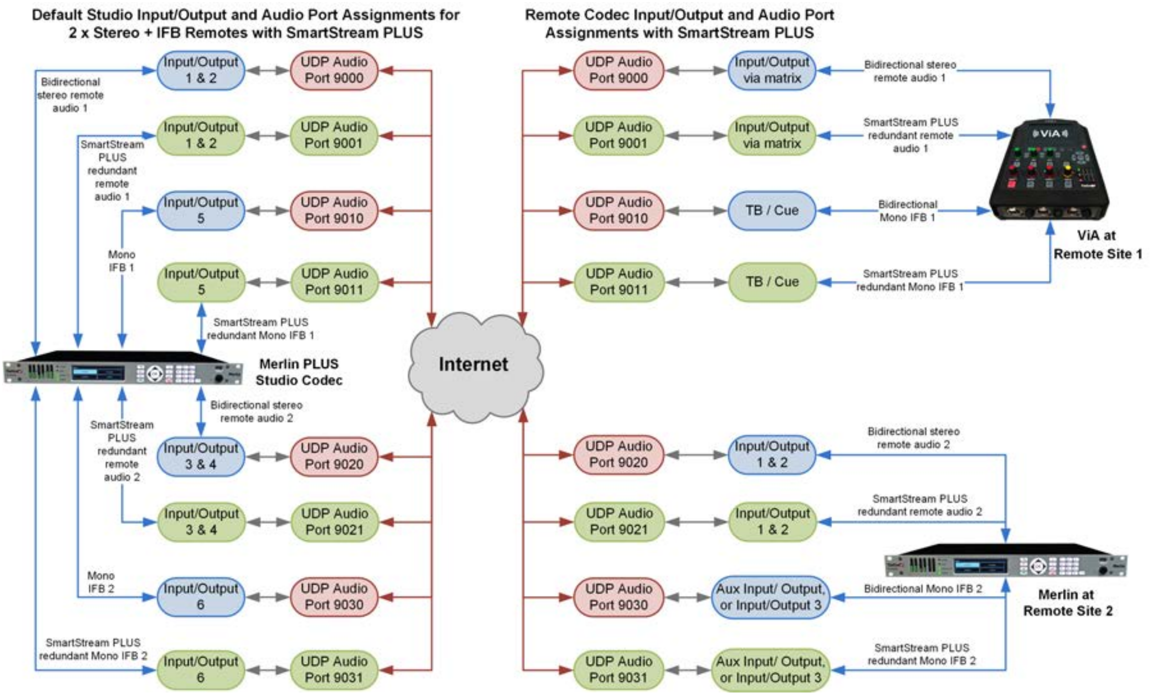
When this program is loaded, any Tieline G3 codec dialing in using IP1 (using default Tieline IP port settings) will be routed to output 1. A codec dialing in using IP2 will be routed to output 2.

19.13 Configure Mono or Stereo + IFB Dialing Programs

This program is designed to allow remote Merlin and Merlin PLUS codecs to dial a Merlin or Merlin PLUS codec at the studio and transmit:

1. A bidirectional mono or stereo audio stream connection.
2. A separate bidirectional mono IFB audio stream for communications.

This program can also include SmartStream PLUS dual IP streaming. The following diagram indicates the default input, output and port assignments for **Mono or Stereo Peer-to-Peer + IFB** Programs using SmartStream PLUS when two remote codecs are dialing a Merlin PLUS codec at the studio.



2 x Mono/Stereo Peer-to-Peer + IFB Remotes dialing Merlin PLUS at the studio

The following setup instructions describe how to configure a stereo audio stream and IFB audio stream, with a backup connection, in order to connect with a Merlin or Merlin PLUS codec at the studio.

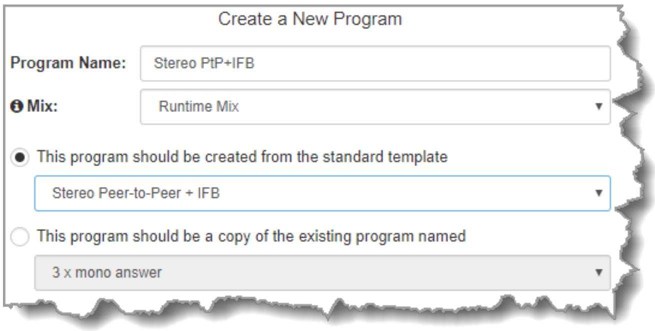
Configuring a Mono or Stereo Audio Stream: Dialing

Important Notes: Before you commence program configuration please note:

- The auxiliary input is used by default for the IFB communications channel in Merlin codecs. In Merlin PLUS codecs the auxiliary input and XLR input 3 are mixed together by default for 1 x Mono/Stereo Peer-to-Peer + IFB programs.
- You cannot edit a program when it is currently 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 type of program. 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 and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.

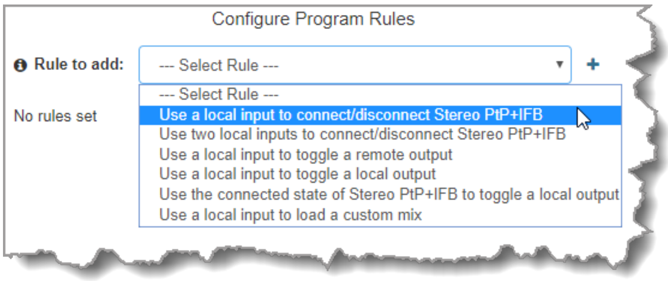
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- POTS is not supported for stereo audio stream connections.
- To learn more about programs see the section titled About Program Dialing.

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 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 + IFB**, or if you want to use an existing program as a template, select this option. Note: The following example is configured to connect a stereo audio stream and mono IFB stream. 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.

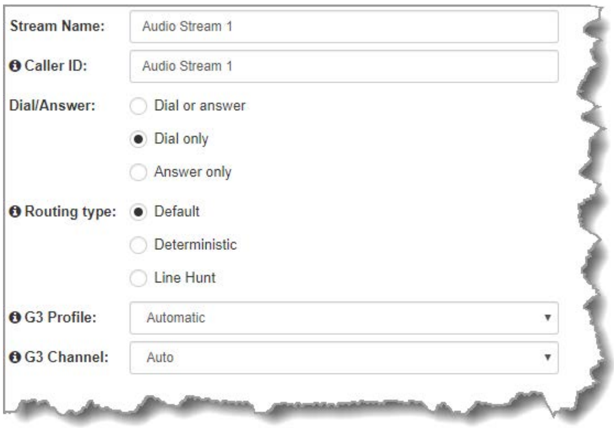


Important Notes for Rules:

- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet 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). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	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 ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	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 when answering a call from a G5 codec.

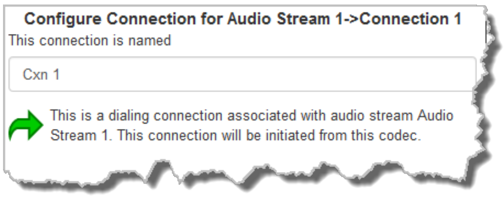
1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. Note: This is overridden in a ViA codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. 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 codec.

4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

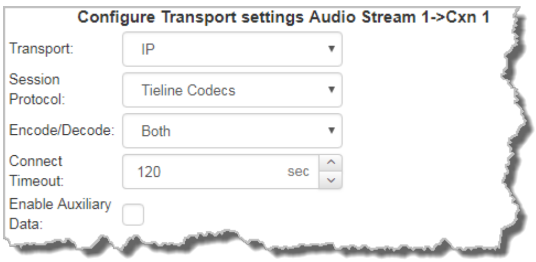
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

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 connection name 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.

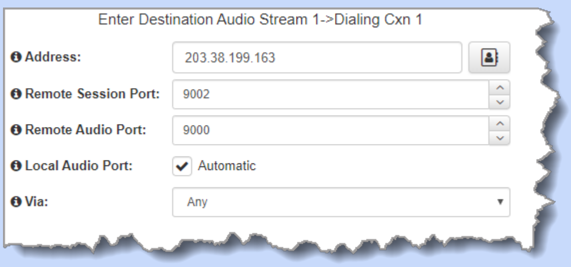


Important Note: See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

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 **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.

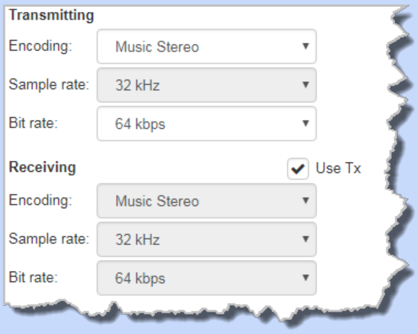


Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), 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. 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**.

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 and configure a failover connection or SmartStream PLUS for this audio stream (recommended).

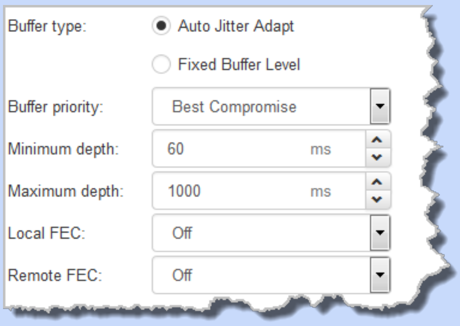
Note: If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio (when creating 2 x Mono or Stereo Peer-to-Peer + IFB connections), use **Remote Audio Port 9020** to configure the second mono/stereo dialing connection at the studio. Mono or stereo program audio over this audio stream connection will be routed via audio inputs/outputs 3 and 4 on the studio Merlin PLUS codec.

When adjusting encoding settings, 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:

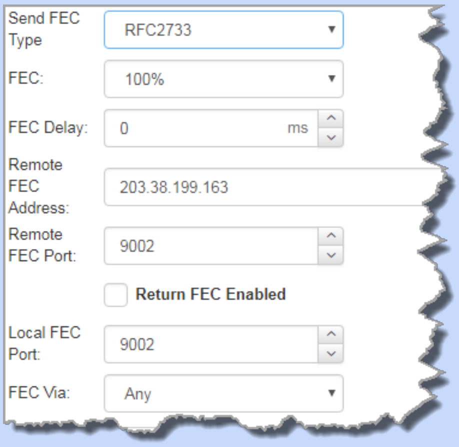
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed 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:

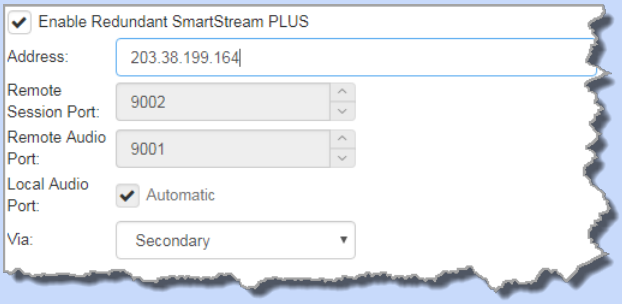
- If you select **Sessionless** or **SIP** as the **Session Protocol** then RFC-compliant FEC is displayed. Configuration instructions are displayed in the right-hand pane.

- **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. Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby an alternative connection is dialed if the primary connection fails.

By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. 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.

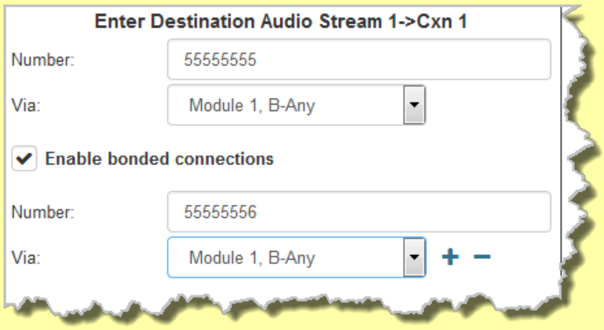


Important Notes:

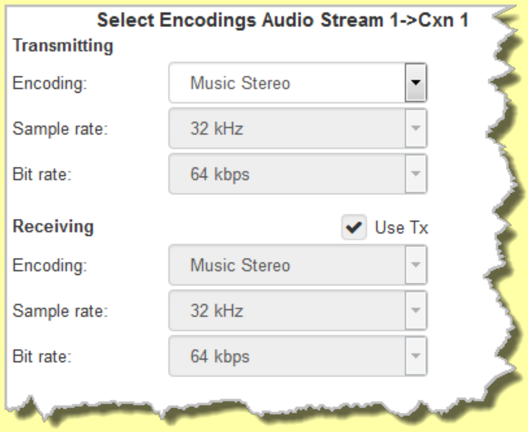
- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three 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/

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.

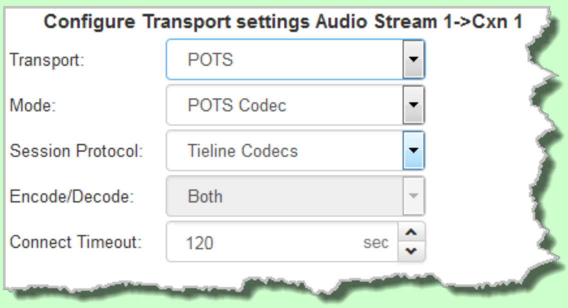


Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).

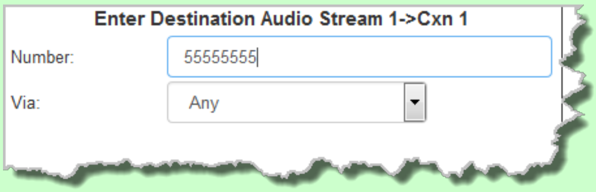


Dialing settings for this ISDN audio stream are now complete.

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2 to** specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



Dialing configuration settings for this POTS audio stream are now complete.

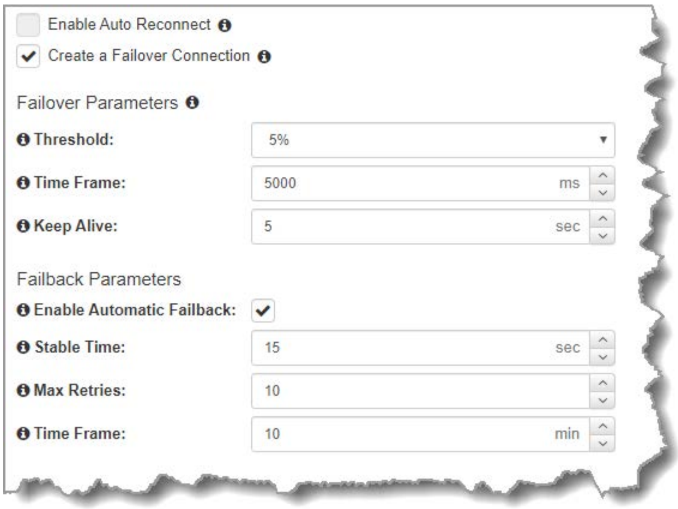
Configuring a Failover Connection or Auto Reconnect

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

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 to select the check-box for **Create a Failover Connection**. Adjust the 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 within the following table can be used to assist with failover connection configuration.

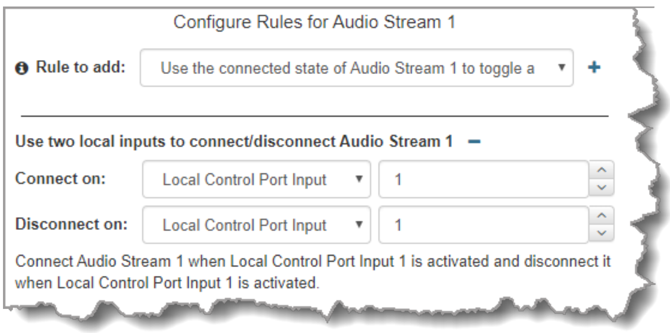
	Screen Display	Description
1	Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame against which lost data is measured
3	Keep Alive	The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity
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	The time frame used to measure the number of fail back retries attempted

2. Enter a name for the backup connection.

3. After configuring all settings there are 2 options:

   i. Click **Next** to configure rules options.
   ii. Click **Next Stream** to configure the second audio stream.

4. To configure new stream related 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.



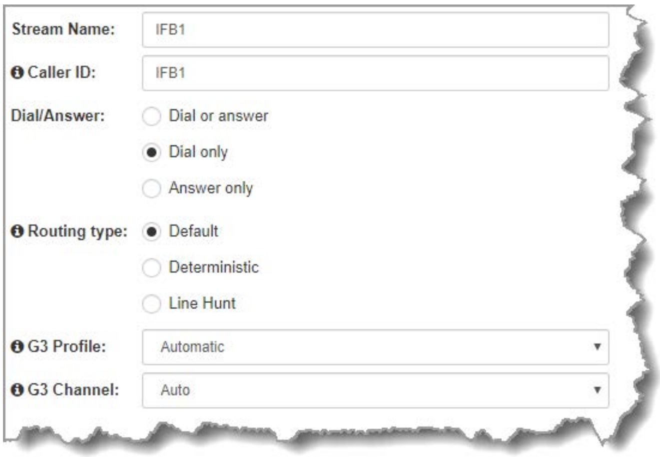
Important Note: Connection-related rules are not displayed in **Answer only** audio streams.

5. Click **Next** to configure the IFB audio stream.

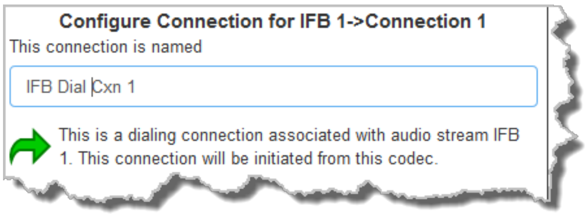
Configure the Bidirectional IFB Audio Stream

When you have finished configuring SmartStream PLUS, Auto Reconnect or a backup connection, proceed with configuration of the IFB audio stream in the wizard.

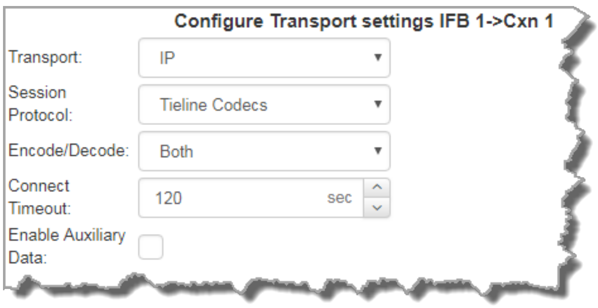
1. Enter the IFB **Stream Name**, add a **Caller ID** and configure the codec to **Dial only**. Then click **Next**.



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



3. Configure the transport settings for the connection, then click **Next**.

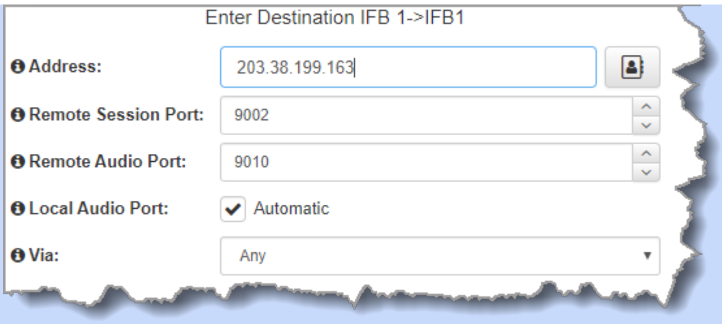


Important Note: See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

4. Configure destination codec dialing and encoding settings:

IP

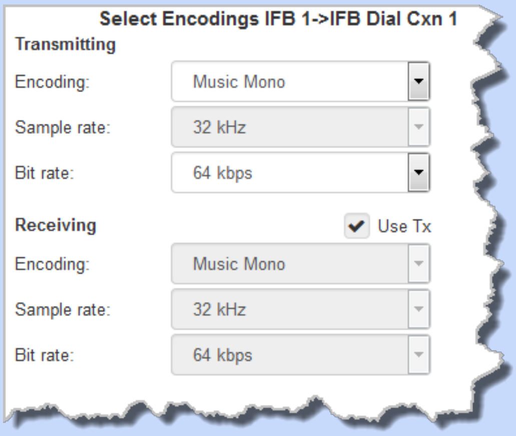
For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **ETH1**) or **Secondary** (port **ETH2**). Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.



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 a backup connection or SmartStream PLUS for this audio stream (recommended).

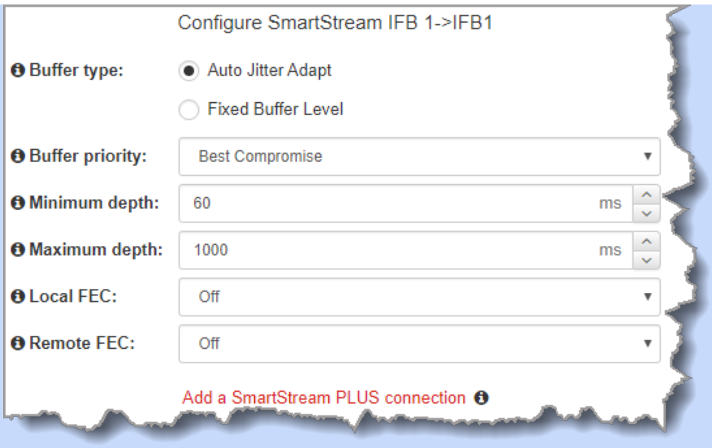
Note: The default **Remote Audio Port** is **9010** for this IP audio stream. If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio (when creating 2 x Mono or Stereo Peer-to-Peer + IFB connections), use **Remote Audio Port 9030** to configure the second IFB connection at the studio. IFB audio over this audio stream connection will be routed via audio input/output 6 on the studio Merlin PLUS codec.

Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.

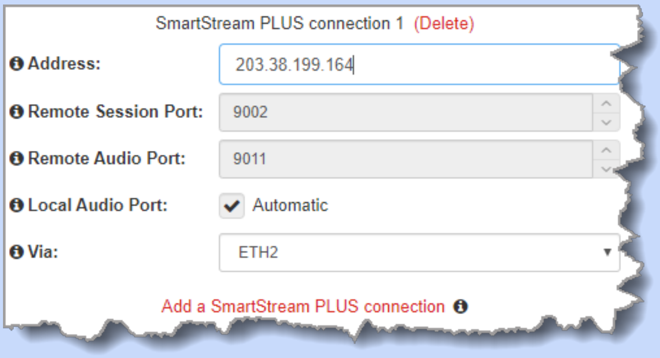


For IP connections click to configure:

- **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.



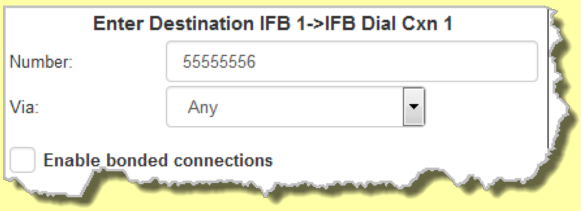
Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.



By default, primary IP streaming is via **ETH1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **ETH2**. The redundant stream uses **Remote Audio Port 9011** by default, and provides automatic IP streaming backup in case one IP connection fails.

ISDN

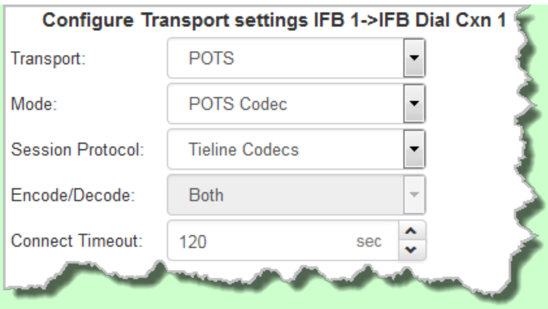
For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.



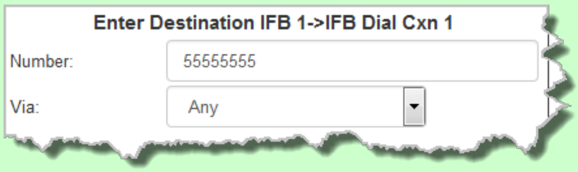
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended). Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2 to** specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



Dialing configuration settings for this POTS audio stream are now complete.

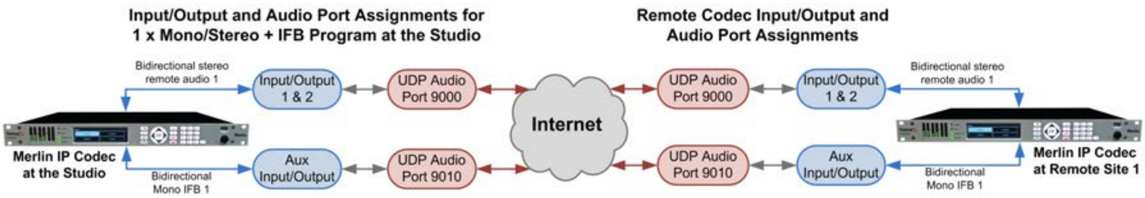
5. Continue through the steps in the wizard to complete configuration in the same way as the first connection was configured. Then click **Save Program** to complete configuration, then click **Finish** to exit the wizard.
6. The newly created program can be loaded from within the **Program Manager panel** 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.

19.14 Configure Mono or Stereo + IFB Answering Programs

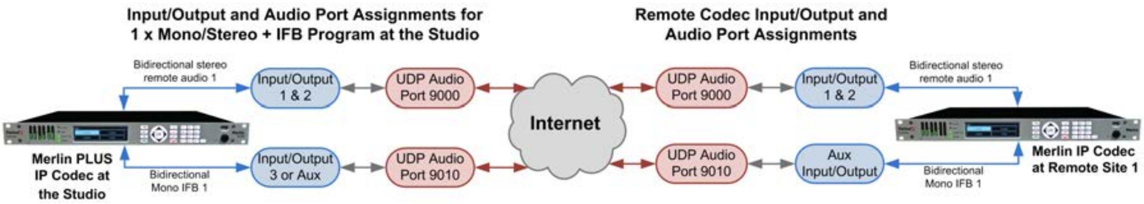
This program is designed to allow Merlin and Merlin PLUS codecs to answer a call from an incoming codec and receive:

1. A bidirectional mono or stereo audio stream connection.
2. A separate bidirectional mono IFB audio stream for communications.

A remote Merlin, Merlin PLUS or ViA codec can dial into a studio Merlin or Merlin PLUS codec to create these audio stream connections.



Merlin codec at the studio connects to a remote Merlin codec



Merlin PLUS at the studio connects to a remote Merlin (Note Aux/input 3 IFB routing option)

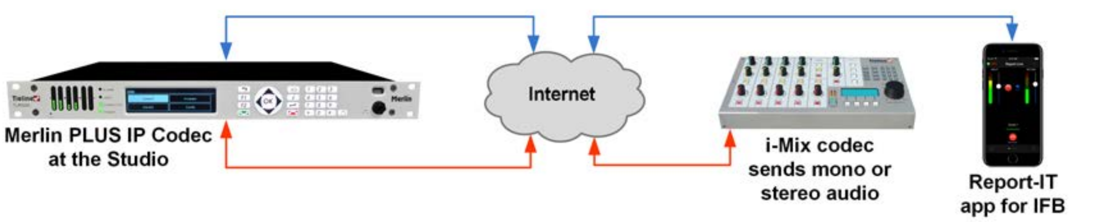
Answering Mono + IFB Calls from Tieline G3 Codecs

The **G3 Mono Peer-to-Peer + IFB** program template can be used to configure a specific jitter buffer setting when answering calls from Tieline Commander and i-Mix G3 codecs which are using a **Man Dflt Mono/IFB** profile on a Merlin or Merlin PLUS. A Merlin codec will automatically detect an incoming **Man Dflt Mono/IFB** call from a G3 codec, but to configure the jitter buffer setting independently requires a **G3 Mono Peer-to-Peer + IFB** answering program to be loaded on the Merlin or Merlin PLUS with appropriate settings. Note: Only one stereo audio stream is connected in this mode and one channel is used for program and the other for communications. This is to interface seamlessly with how G3 codecs are configured.

Connecting with Two Separate Codecs

A Merlin or Merlin PLUS codec can accept 2 calls to create a **Mono or Stereo Peer-to-Peer + IFB** program. For example, an i-Mix G3 could dial and transmit stereo program and a second IP codec or smartphone running the Report-IT application can connect and deliver a bidirectional mono IFB audio stream.

1 x Mono or Stereo Peer-to-Peer + IFB via Two Remote Devices



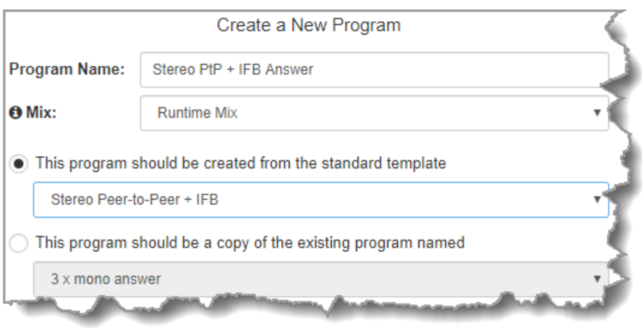
Important Note: Remember to lock the program in the studio codec when connecting a **Mono or Stereo Peer-to-Peer + IFB** program using two devices at the remote site. This will avoid the first mono or stereo call unloading the **Peer-to-Peer + IFB** program at the studio and loading a mono or stereo peer-to-peer program, which would cause the second connection to fail.

Configuring Mono or Stereo Peer-to-Peer + IFB Programs

In most situations the studio codec will answer incoming audio stream connections from the remote site. The following procedure outlines configuration of an answering program for the studio codec.

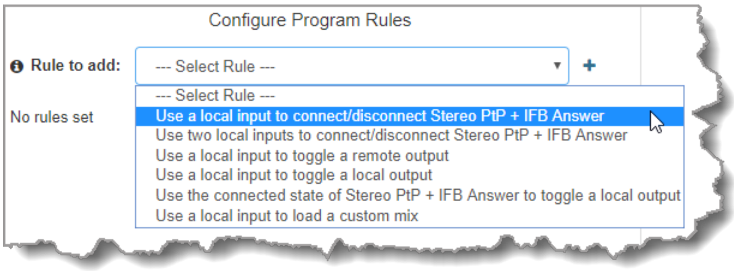
Important Notes: Before you commence program configuration please note:
- The auxiliary input is used by default for the IFB communications channel in Merlin codecs. In Merlin PLUS codecs the auxiliary input and XLR input 3 are mixed together by default for 1 x Mono/Stereo + IFB programs.
- You cannot edit a program when it is currently 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 type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- If the codec at the studio will receive both mono and stereo peer-to-peer + IFB calls from different remote sites at different times, we recommend you configure and load a **1 x Stereo Peer-to-Peer + IFB** answering program and lock this in the codec at the studio. This will accept both mono and stereo audio stream connections. If a codec with a **Mono Peer-to-Peer + IFB** program calls the studio, the incoming mono stream will be mixed to both the left and right outputs at the studio.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- POTS is not supported for stereo audio stream connections.
- To learn more about programs see the section titled About Program Dialing.

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 + IFB**, 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.

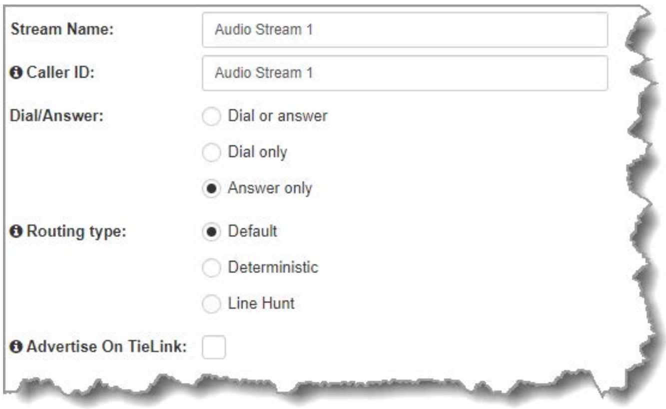
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Notes for Rules:

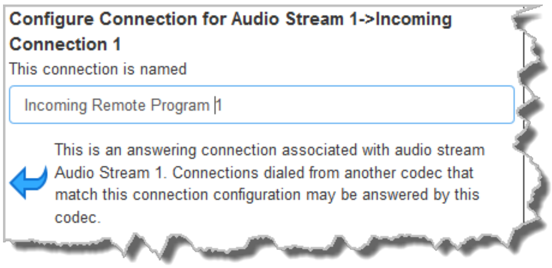
- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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 **Answer only**. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	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 ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	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.

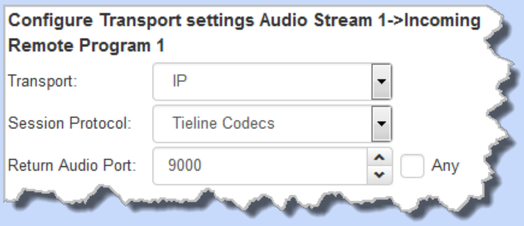
5. Enter the connection name in the text box, then click **Next.**



6. Configure the transport settings:

IP

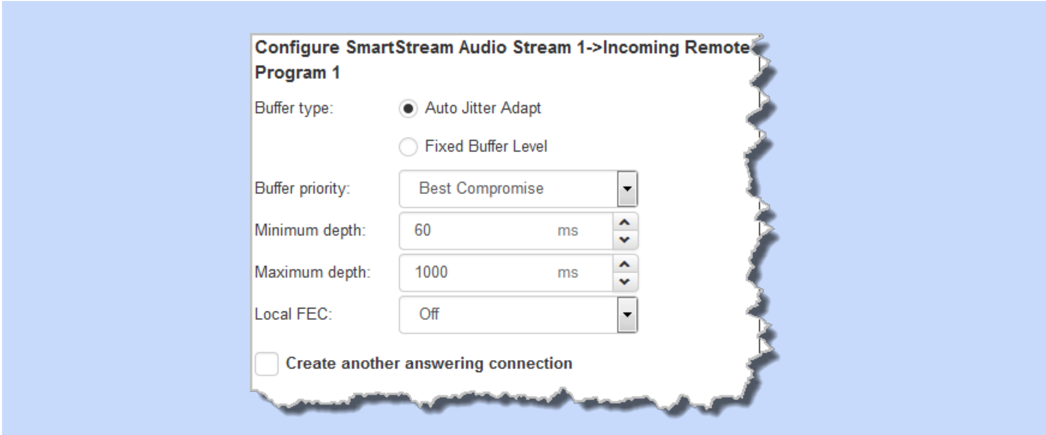
For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected.



Important Note: The **Local Audio Port** is the port 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 **Local Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream. Click to deselect the **Any** check-box to adjust this setting.

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.



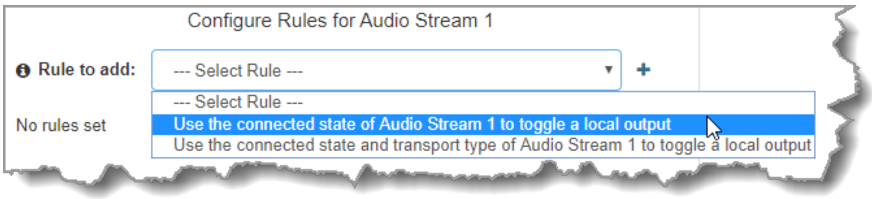
ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

POTS settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

7. After configuring all settings there are 3 options:

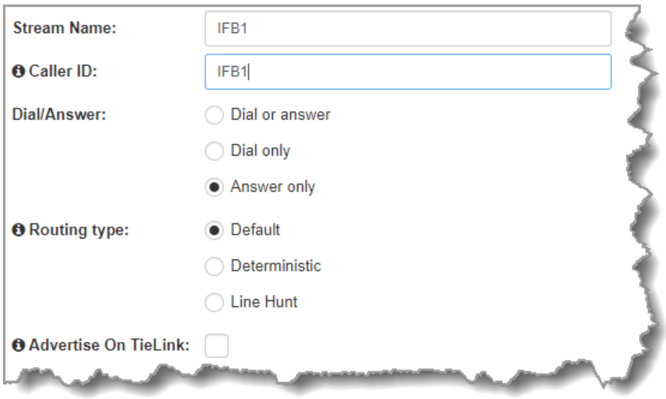
    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
    ii. Click **Next** to configure rules options.
    iii. Click **Next Stream** to configure the IFB audio stream.

8. To configure new stream related 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.

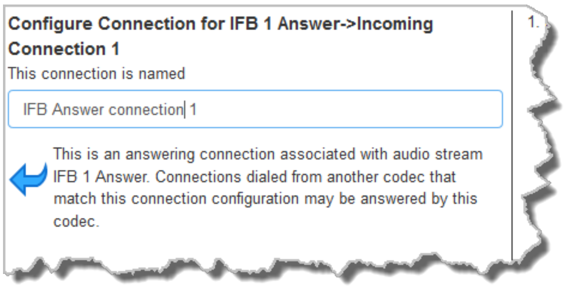


Important Notes:
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

9. Click **Next** to configure the IFB audio stream. Enter the IFB Audio **Stream Name**, add a **caller ID** and configure the codec to **Answer only**. Then click **Next**. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.

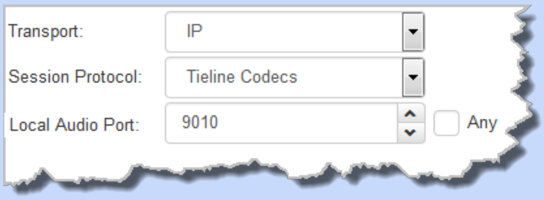


10. Enter the IFB audio stream connection name in the text box and click **Next**.



11. Configure transport settings:

IP: For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.



Important Note: When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Local Audio Port** is automatically configured as UDP audio port 9010 by default for the second audio stream.

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.

ISDN: ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

POTS: POTS settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

12. After configuring all settings there are 3 options:

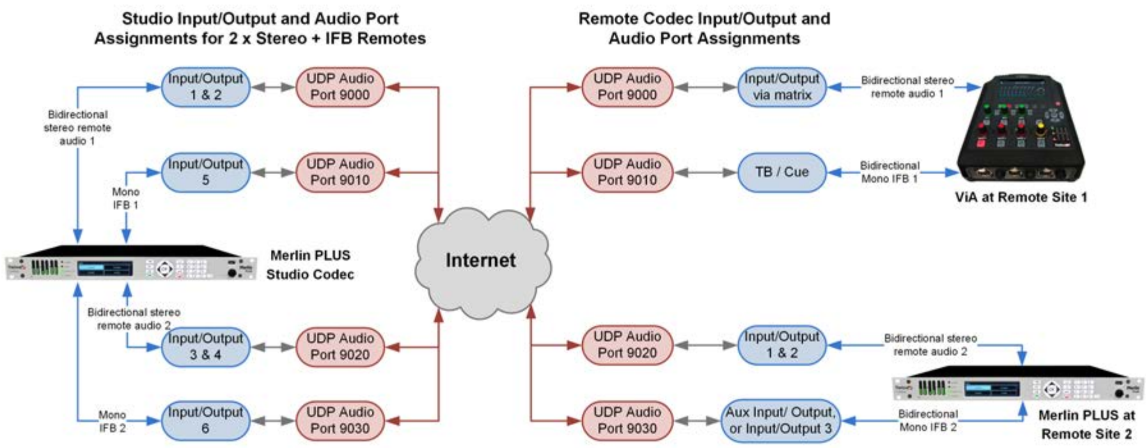
i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
ii. Click **Next** to configure audio stream rules.
iii. Click **Save Program** to save all program settings, then click **Finish** to exit the wizard.

The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**. This program will now allow incoming codec calls to establish **1 x Stereo Peer-to-Peer + IFB** connections and stream audio according to the input/output and port assignments indicated at the beginning of this section.

19.15 Configure 2 Mono or Stereo + IFB Answering Programs

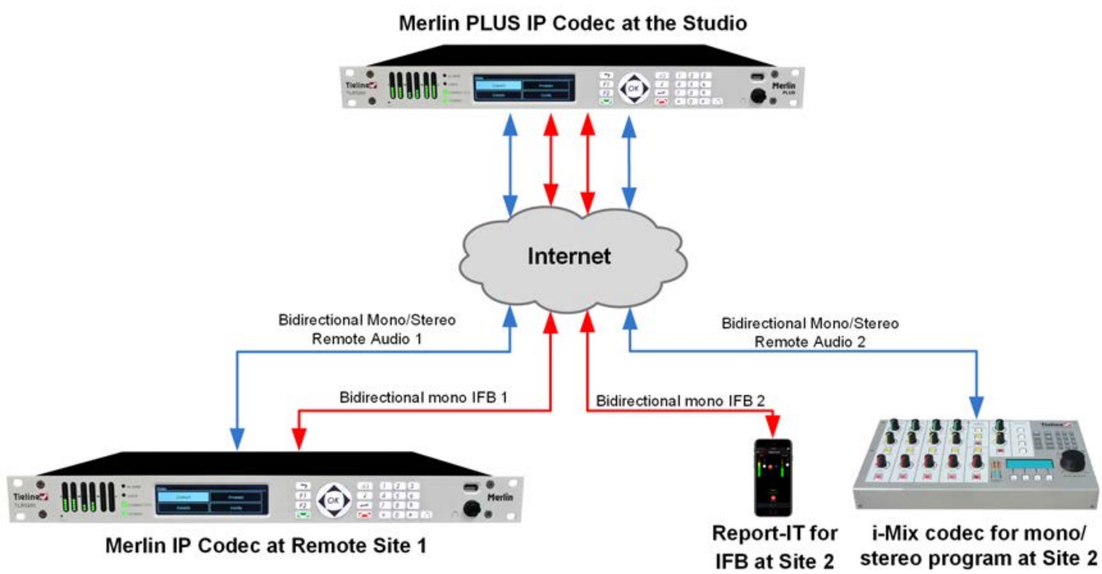
This program is designed to allow a Merlin PLUS codec at the studio answer calls from two incoming ViA, Merlin or Merlin PLUS codecs and receive:

1. Two bidirectional mono or stereo audio stream connections.
2. Two separate bidirectional mono IFB audio streams for communications.



Connecting other Tieline Codecs

When a codec which only supports a mono or stereo audio stream attempts to connect (e.g. Commander G3, i-Mix-G3 or Bridge-IT codecs not supporting a separate IFB audio stream), a Merlin PLUS codec at the studio will accept the call and stream mono or stereo audio only. A second IP codec or smartphone running the Report-IT application can also be configured to connect and deliver the bidirectional mono IFB audio stream.



Important Note: Remember to lock the program when connecting a **Mono or Stereo Peer-to-Peer + IFB** program using two devices at the remote site. This will avoid the first mono or stereo call unloading the **Peer-to-Peer + IFB** program at the studio and loading a mono or stereo peer-to-peer program, which would cause the second connection to fail.

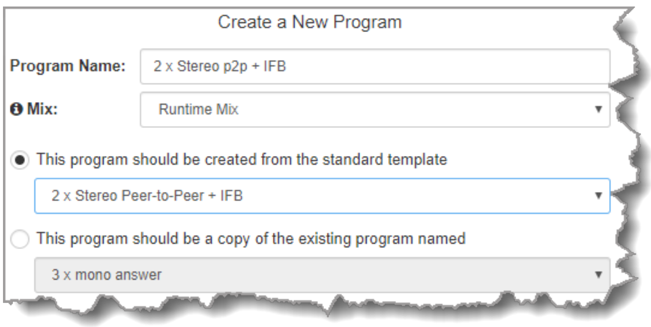
Configuring 2 x Mono or Stereo Peer-to-Peer + IFB Programs

In most situations the studio codec will answer incoming audio stream connections from 2 remote sites. As a result, the following procedure explains configuration of the answering connections required at studio codec.

Important Notes: Before you commence program configuration please note:

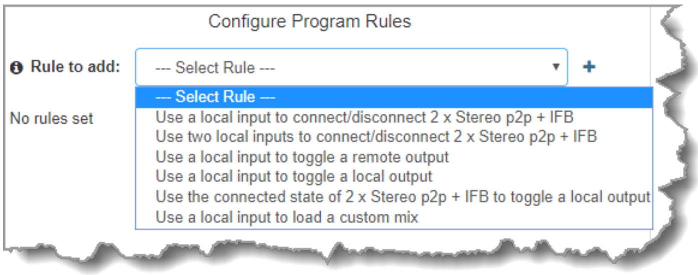
- The auxiliary input is used by default for the IFB communications channel in Merlin codecs.
- You cannot edit a program when it is currently loaded in the codec.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- If your Merlin PLUS codec at the studio will receive both mono and stereo peer-to-peer + IFB calls from different remote sites at different times, we recommend you configure and load a **2 x Stereo Peer-to-Peer + IFB** answering program and lock this in the codec at the studio. This will accept both mono and stereo audio stream connections. If a codec with a **Mono Peer-to-Peer + IFB** program calls the studio, the incoming mono stream will be mixed to both the left and right outputs at the studio.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- POTS is not supported for stereo audio stream connections.
- To learn more about programs see the section titled About Program Dialing.

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 **2 x Mono/Stereo Peer-to-Peer + IFB**, 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.

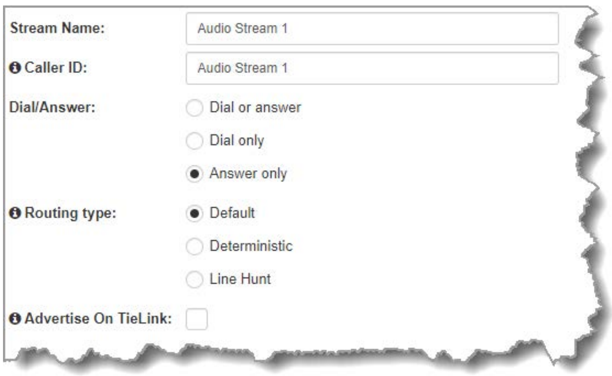
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Notes for Rules:

- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet 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). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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.

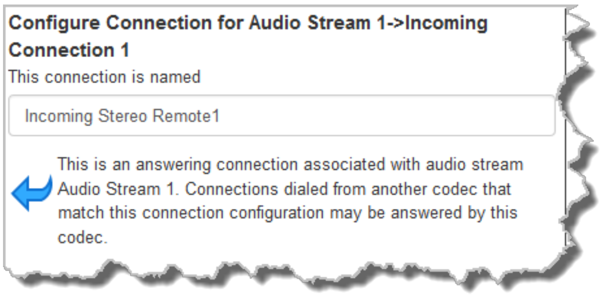
4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to **Answer only**. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	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 ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This

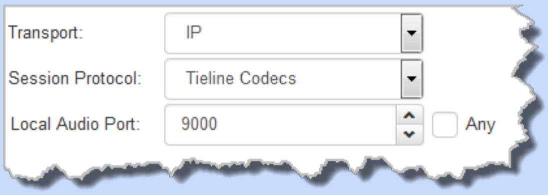
	effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt 	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

5. Enter the name of the connection in the text box, then click **Next.**



6. Configure the transport settings:

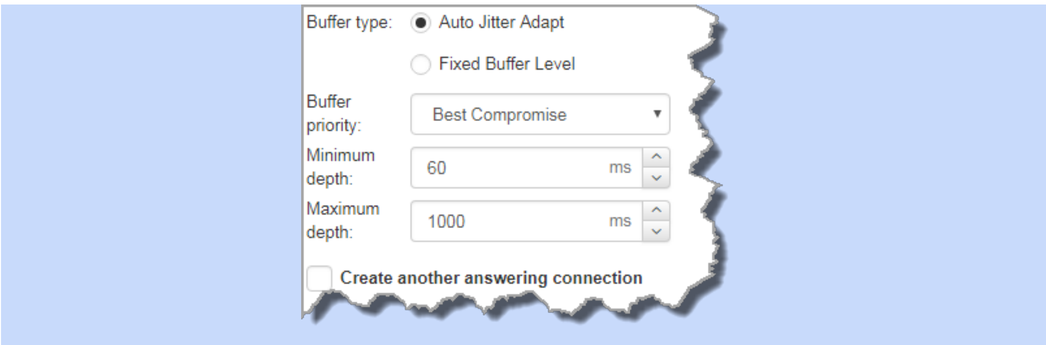
For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected.



Important Note: The **Local Audio Port** is the port 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 **Local Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream. Click to deselect the **Any** check-box to adjust this setting.

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.



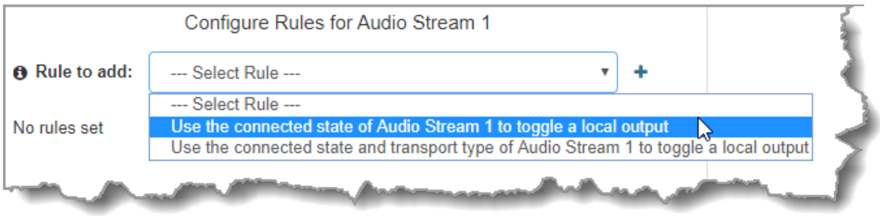
ISDN For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

POTS For POTS, settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

7. After configuring all settings there are 3 options:

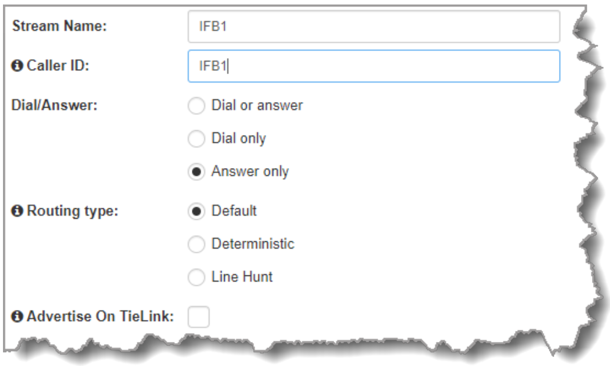
    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
    ii. Click **Next** to configure rules options.
    iii. Click **Next Stream** to configure the IFB audio stream.

8. 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.

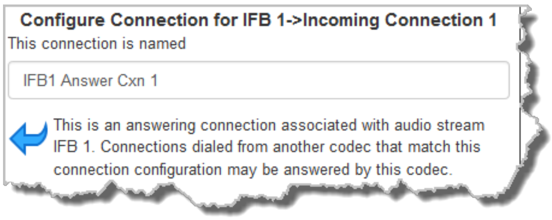


Important Notes:
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

9. Click **Next** to configure the IFB audio stream. Enter the IFB Audio **Stream Name**, add a **caller ID** and configure the codec to **Answer only**. Then click **Next**. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.

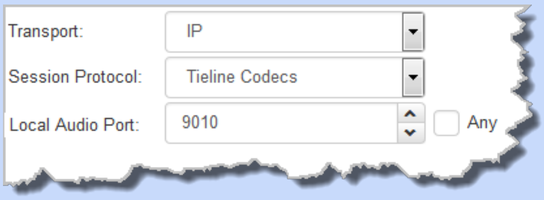


10.Enter the name of the second answering stream connection in the text box, then click **Next.**



11. Configure transport settings:

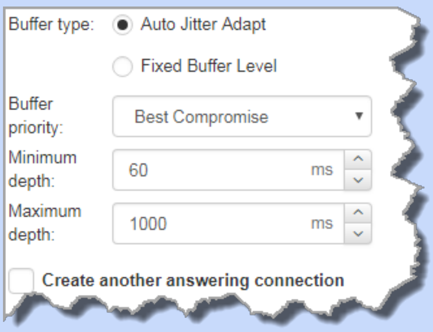
IP For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.



Important Note: When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Local Audio Port** is automatically configured as UDP audio port 9010 by default for the second audio stream.

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.



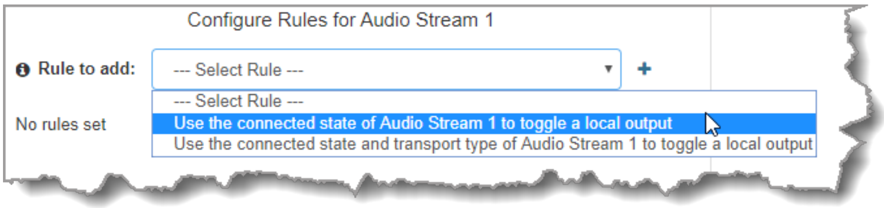
ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

POTS settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

12. 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 **Next** to configure rules options.
    iii. Click **Next Stream** to configure the second stereo audio stream and IFB stream.

13. To configure new stream related 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.

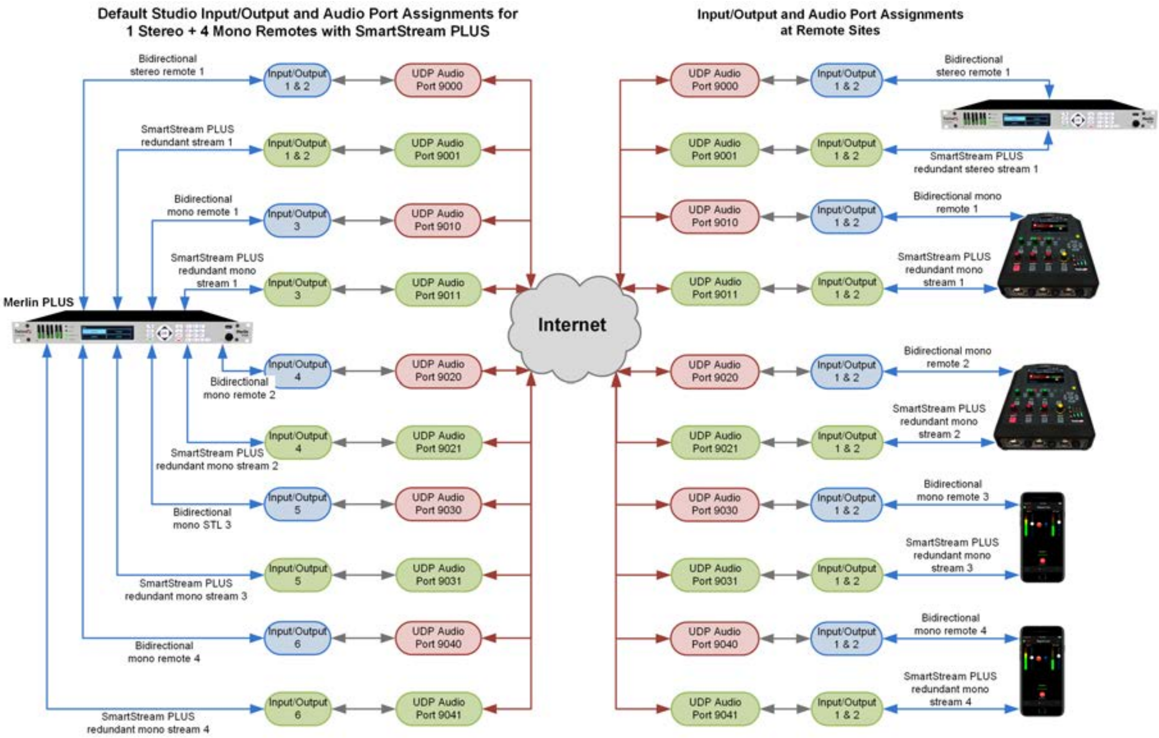


Important Notes:
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

14. Continue through the steps in the wizard to complete configuration of the second program audio stream and second IFB audio stream, in the same way as the first audio streams were configured. Click **Save Program** at the end of this process and then click **Finish**. This program will now allow incoming codec calls to establish **2 x Mono/Stereo Peer-to-Peer + IFB** connections and stream audio as per the input/output and port assignments indicated in the image at the beginning of this section. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**.

19.16 Configure Stereo + 4 x Mono Peer-to-Peer Dialing Programs

This program is allows a Merlin PLUS codec at the studio to answer one stereo and four mono bidirectional audio streams.



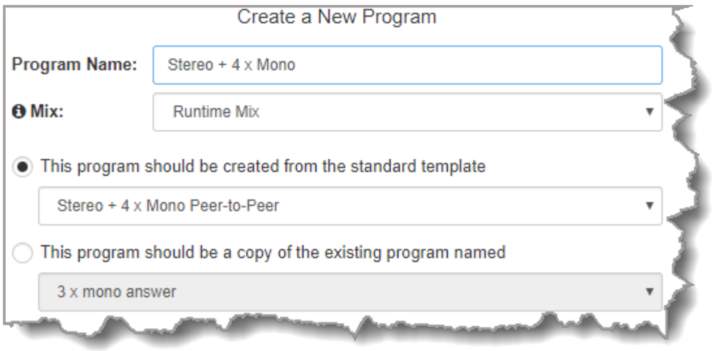
Configuring Stereo + 4 x Mono Peer-to-Peer Programs

Important Notes: Before you start program configuration please note:

- You cannot edit a program when it is currently 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 type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- Session port 9002 is used by default for all audio streams. If you connect to a G3 codec using IP2 you will need to change the session port on the Merlin PLUS codec to 9012 for the second audio stream connection, as this is used by default on G3 codecs for IP2.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled About Program Dialing.
- The stereo audio stream is configured first and then the four mono audio streams.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- POTS is not supported for stereo audio stream connections.
- Selection of a mono algorithm and associated settings is the only difference when comparing configuration of a mono audio stream to a stereo audio stream.

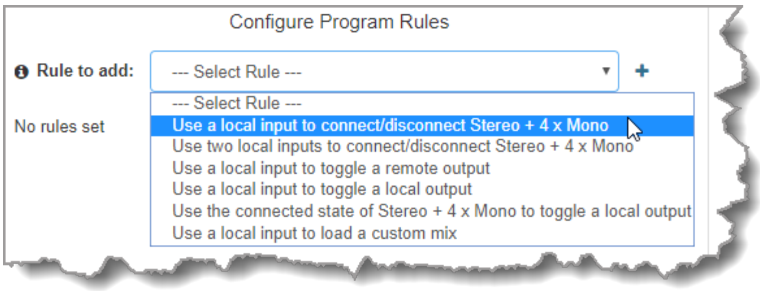
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:

- 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 **Stereo + 4 x Mono 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.

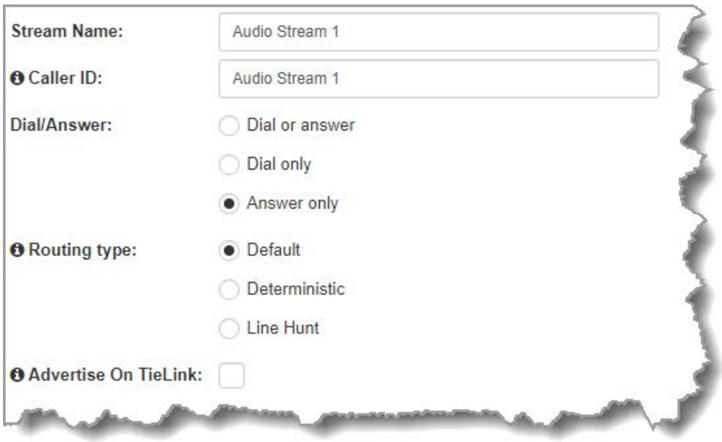
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



Important Notes for Rules:

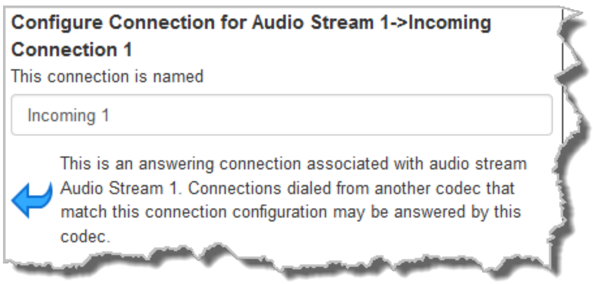
- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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 **Answer only**. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.

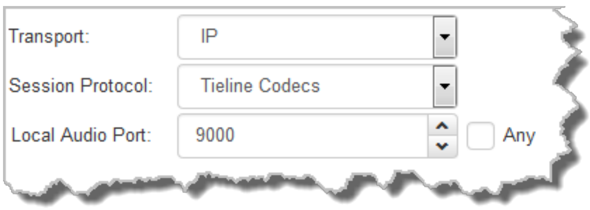


Routing Type Options:	
Default ISDN POTS	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic IP ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	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.

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



6. Configure the transport settings for the connection, then click **Next**.

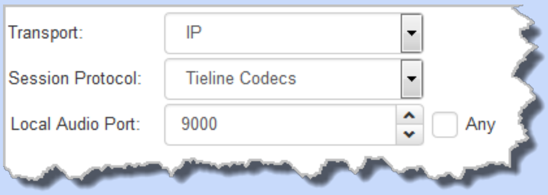


Important Note: See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure the transport settings:

IP

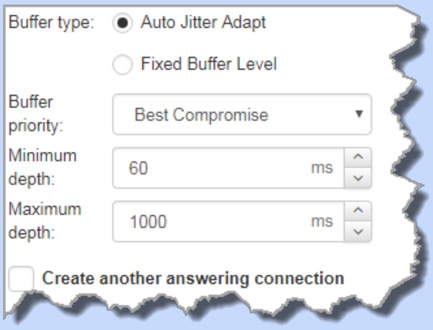
For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected.



Important Note: The **Local Audio Port** is the port 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 **Local Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream. Click to deselect the **Any** check-box to adjust this setting.

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.



ISDN

For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

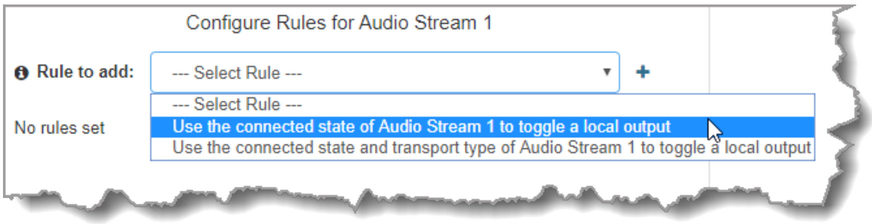
POTS

For POTS, settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

8. 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 **Next** to configure rules options.
   iii. Click **Next Stream** to configure the second audio stream.

9. 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.

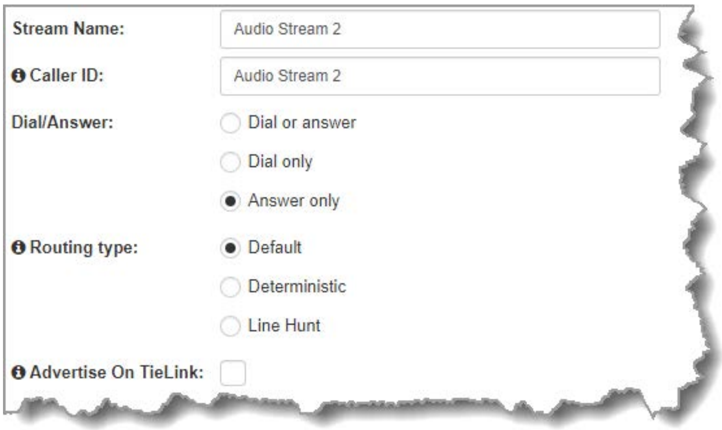


Important Note: Connection-related rules are not displayed in **Answer only** audio streams.

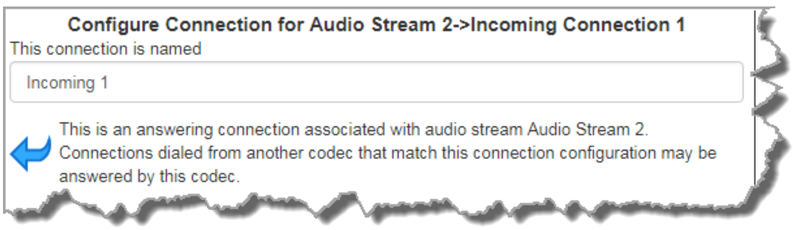
10. Click **Next** to configure the next audio stream.

Configure the Second Peer-to-Peer Audio Stream

1. Enter the second **Stream Name**, add a **caller ID** and configure the program to **Answer only**. Then click **Next**. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



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

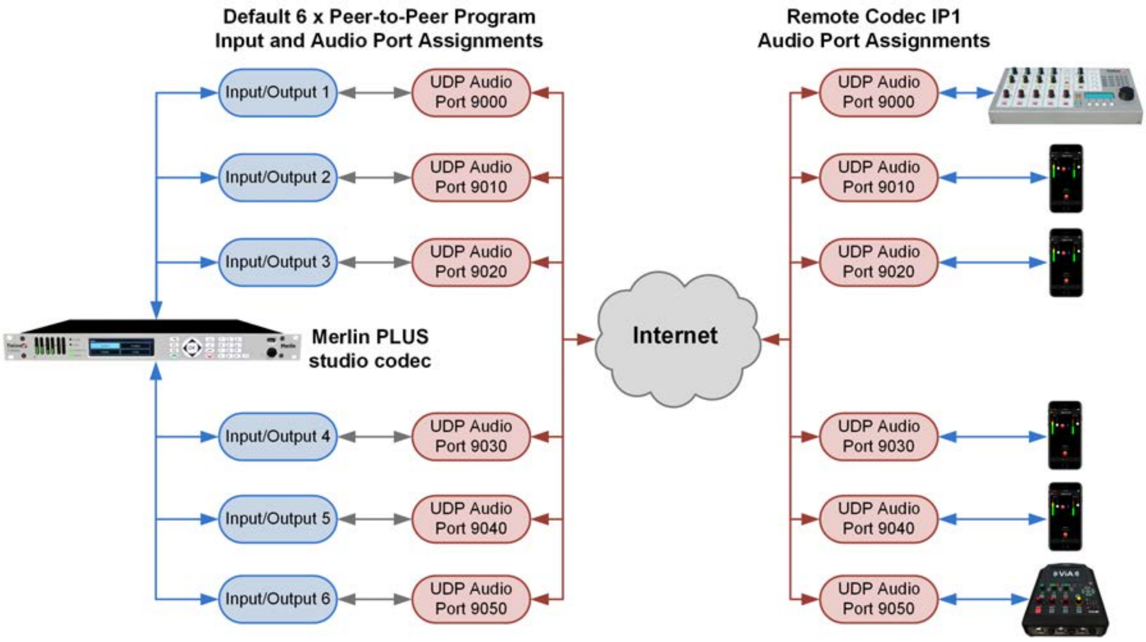


3. Continue through the steps in the wizard and complete configuration of the second audio stream (first mono audio stream) in the same way as the first audio stream was configured. Configure the remaining mono audio streams in the same way. Click **Save Program** at the end of this process, then click **Finish** to exit the wizard. Note: Load and lock this custom program to ensure it cannot be unloaded by a codec dialing in with a different program type.

This program will allow incoming codec calls to establish **Stereo + 4 x Mono Peer-to-Peer** connections and stream audio as per the input/output and port assignments indicated in the image at the beginning of this section. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**.

19.17 Configure 6 x Mono Peer-to-Peer Answering Programs

It is possible for a Merlin PLUS codec at the studio to create six simultaneous bidirectional mono peer-to-peer audio stream connections with different remote codecs or smartphones using the Report-IT app. The following image displays the default audio port settings used when creating a 6 x Mono Peer-to-Peer Program. Remote codecs configured with corresponding audio port settings will route audio to specific inputs/outputs of the Merlin PLUS codec as shown.



Select the **6 x Mono Peer-to-Peer** option from the drop-down menu in the first screen of the program wizard to configure a program allowing up to six mono audio stream connections with different IP codecs or smartphones using Report-IT.

In nearly all situations the studio codec will receive all calls. As a result, the following program wizard procedure displays the various configuration screens to configure an answering connection for each incoming call from up to 6 different devices.

Both mono and stereo audio stream connections can be answered by a Merlin PLUS codec when using a **6 x Mono Peer-to-Peer** program. To do this load and lock a **6 x Mono Peer-to-Peer** answering program in a Merlin PLUS codec at the studio. If a codec using a stereo program attempts to connect then the studio codec will accept the call and mix the two incoming audio channels into a single audio channel to allow a mono peer-to-peer connection.

See Configuring Merlin Point-to-Point Programs for more details about individual settings within the program wizard.

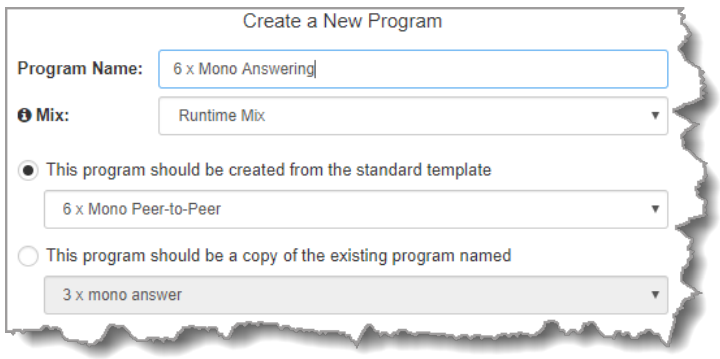
Routing 6 Incoming Mono Audio Streams to Specific Codec Outputs

Important Notes: Before you start program configuration please note:

- You cannot edit a program when it is currently 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 type of program. 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 and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- To learn more about programs see the section titled About Program Dialing.
- 6 audio streams must be configured.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.

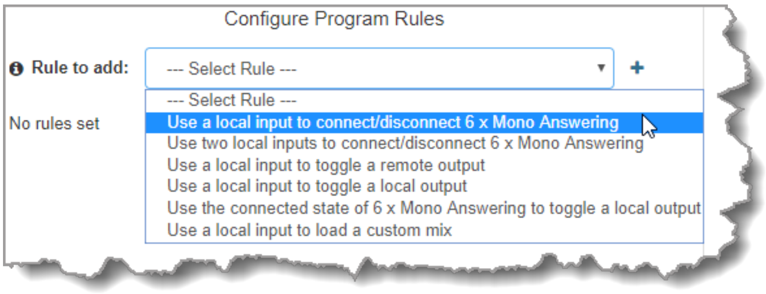
If your intention is to ensure 6 incoming mono peer-to-peer audio stream connections are always routed to the same outputs, configure a new answering program as follows:

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 **6 x Mono 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.

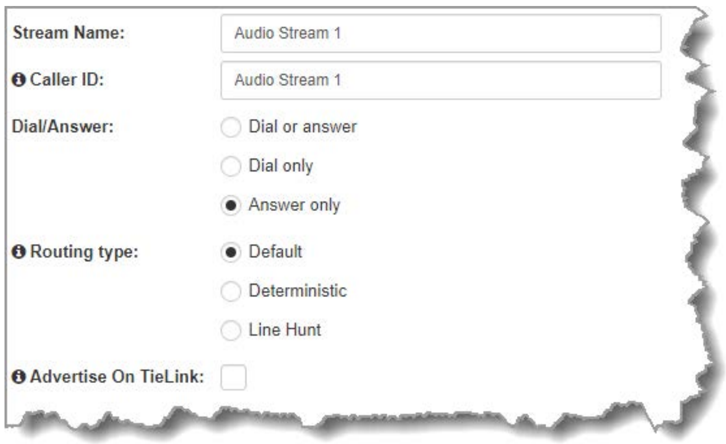
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



Important Notes for Rules:

- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet 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). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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.

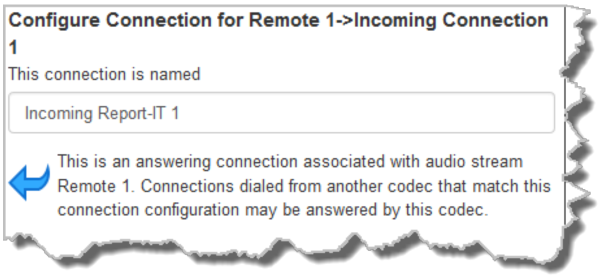
4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to **Answer only**. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	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.

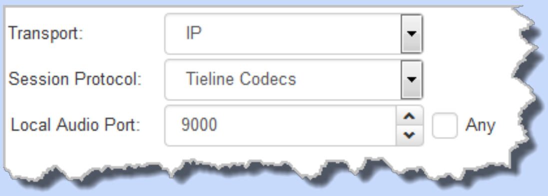
Deterministic IP ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	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.

5. Enter the name of the connection in the text box, then click **Next**.



6. Configure the transport settings:

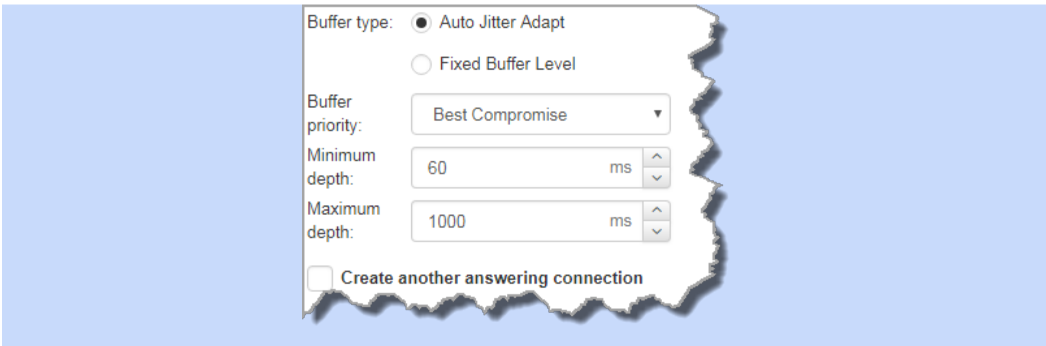
IP For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.



Important Note: 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 **Local Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream.

Note: Deselecting the **Any** check-box ensures that any codec connecting using the audio port specified will always be routed to codec output 1. By default, port 9000 is the port specified for the first IP audio stream connection made by any Tieline codec; port 9010 is used by default for a second IP connection. Merlin PLUS default ports for mono connections 3 to 6 are 9020, 9030, 9040 and 9050. If you need to change this setting on your remote dialing codec, or in your Report-IT smartphone application, this will need to be adjusted accordingly in the answering codec.

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.



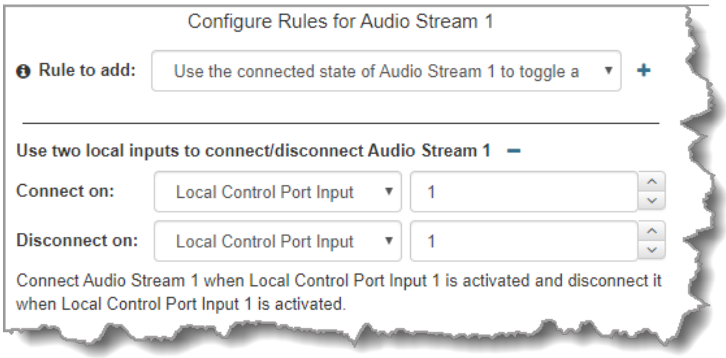
ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

POTS settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

7. After configuring all settings there are 3 options:

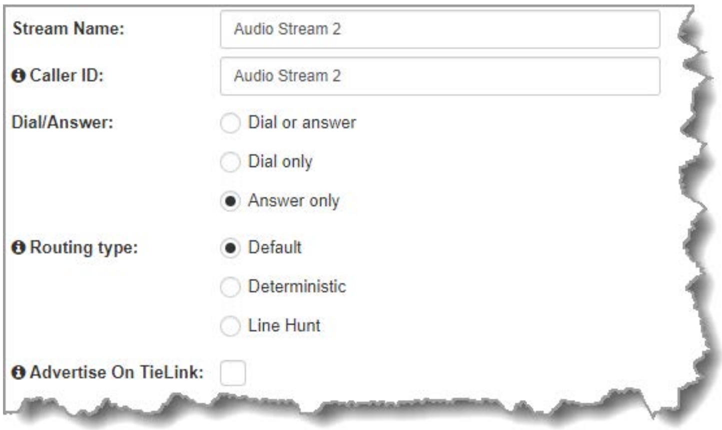
    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
    ii. Click **Next** to configure rules options.
    iii. Click **Next Stream** to configure the next audio stream's settings.

8. 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.

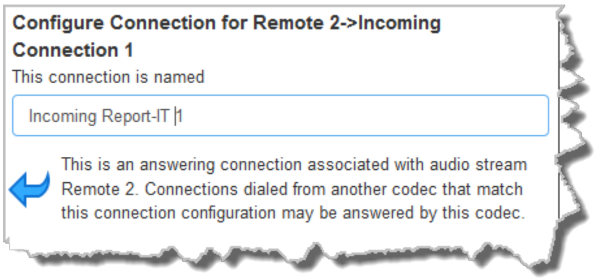


Important Notes:
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

9. Click **Next** to enter the second **Stream Name**, add a **Caller ID** and configure the codec to **Answer only**. Then click **Next**.



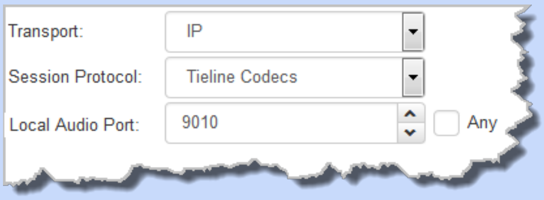
10. Enter the name of the audio stream connection in the text box and click **Next**.



11. Configure the transport settings for this connection:

IP

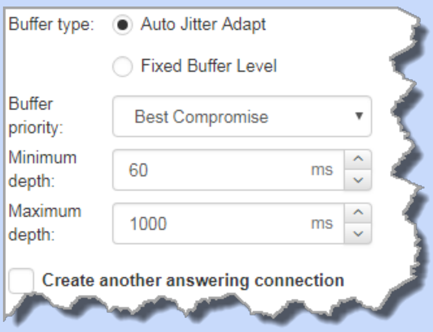
For IP click the drop-down **Session Protocol** menu and select **Tieline Codecs** and ensure the **Any** check-box is not selected, then click **Next**.



Local Audio Port 9010 is used by default for a second IP 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.



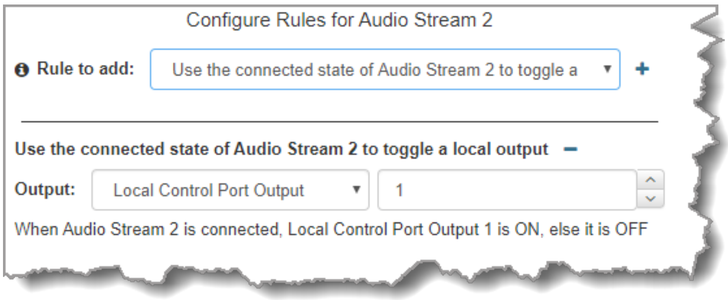
For ISDN settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

For POTS, settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

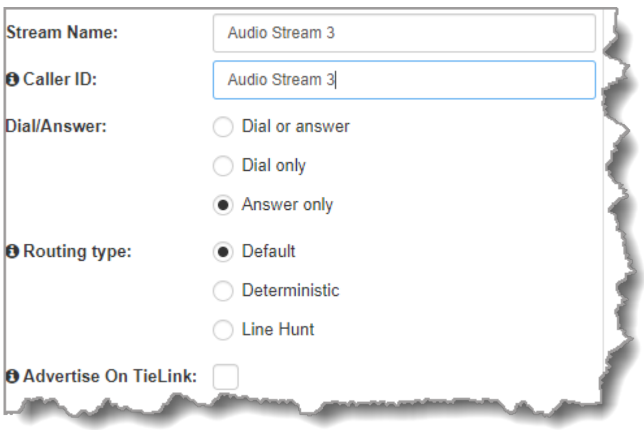
12. After configuring all settings there are 3 options:

    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 **Next** to configure rules options.
    iii. Click **Next Stream** to configure the next audio stream's settings.

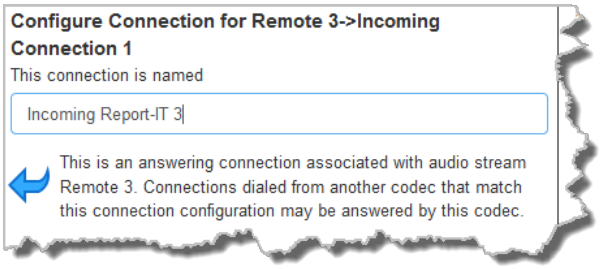
13. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



14. Click **Next** to configure the next **Stream Name**, add a **Caller ID** and configure the codec to **Answer only**. Then click **Next**. Select **Advertise On TieLink** when configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



15. Enter the name of the audio stream connection in the text box and click **Next**.



16. Continue through the program wizard and configure all 6 connections, then click **Save Program** to save program settings and click **Finish** to exit the wizard.

Important Note: The default Merlin PLUS answering program uses audio port 9020 for a third audio stream IP connection. Merlin PLUS default ports for mono connections 3 to 6 are 9020, 9030, 9040 and 9050. If you need to change this setting on your dialing codec, or in your Report-IT smart phone application, this will need to be adjusted accordingly in the answering codec.

17. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**.

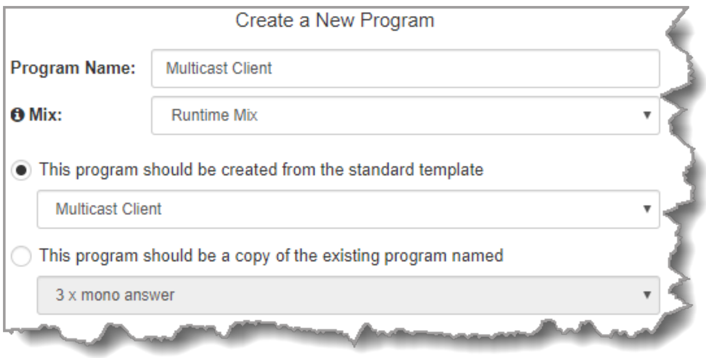
19.18 Configure Multicast Client Programs

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.
- 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.
- 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.
- To learn more about programs see the section titled About Program Dialing.
- 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 the first multicast, 9010 for the second multicast and 9020 for the third multicast. The client and server port settings must match to receive an audio stream. E.g. if a client codec wishes to receive multicast audio stream 2 then it must use audio port 9010.

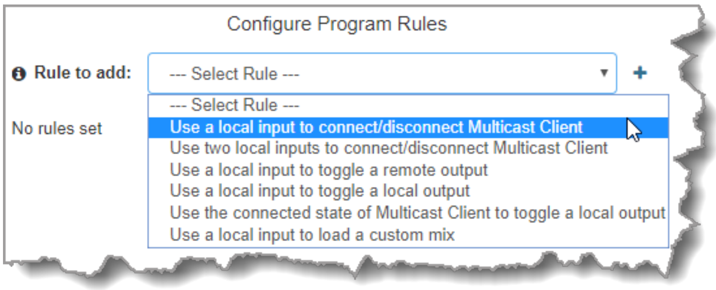
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data.
  - It is not possible to connect to a G3 codec and receive multicast IP audio streams.
  - To copy multicast client programs onto multiple codecs see Backup and Restore Functions.

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.

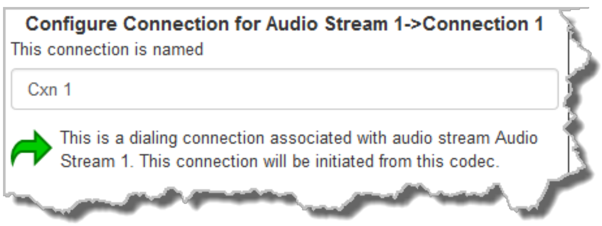


Important Notes for Rules:

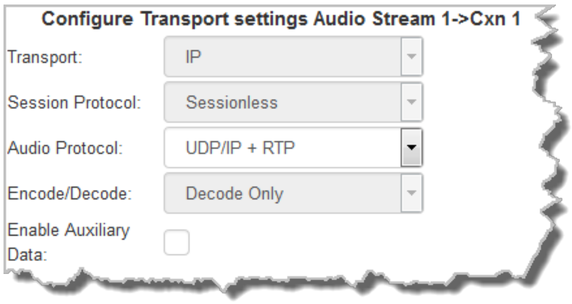
- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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**, 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**.

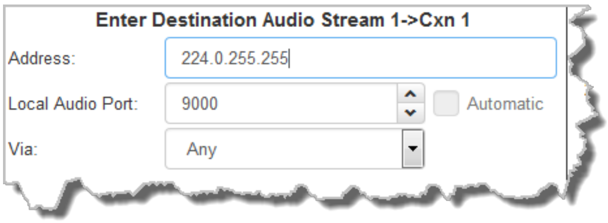


6. Configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC compliant streaming.

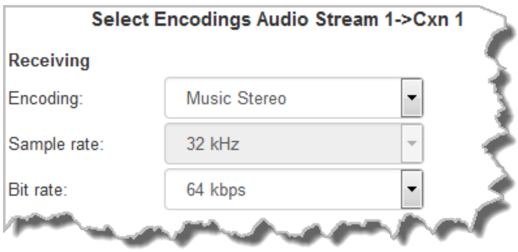


Important Note: See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

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 **ETH1**) or **Secondary** (port **ETH2**), then click **Next**. Note: By default **Any** will select **ETH1** if it is available and **ETH2** if it is unavailable.

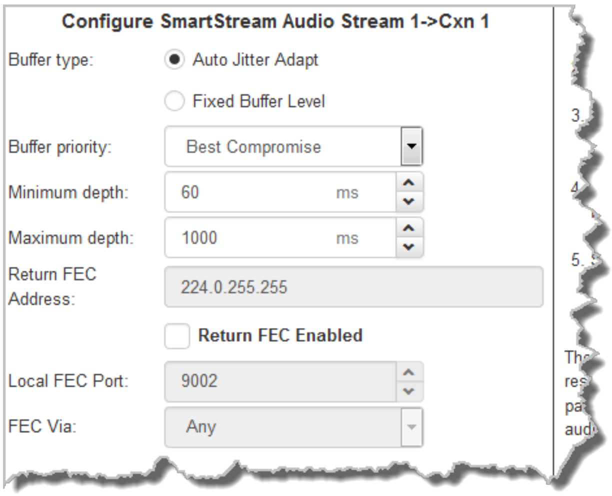


8. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.



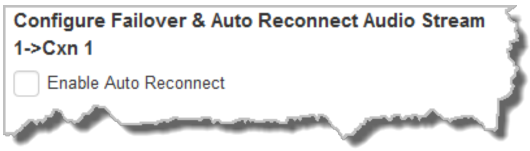
9. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
   - 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**.



Important Notes: 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.

10. Click the **Enable Auto Reconnect** check-box to turn this feature on, then click **Next**.

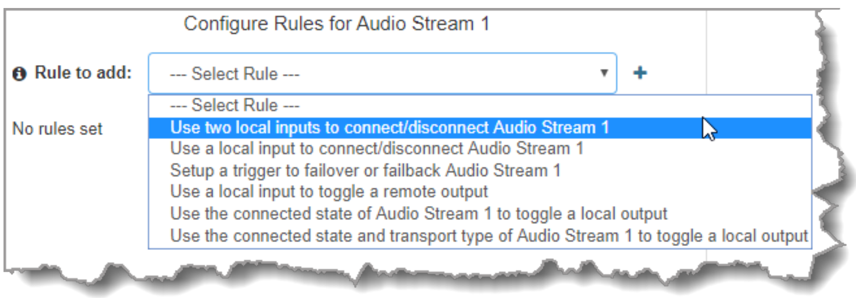


11. After configuring these settings there are 2 options:

   i. Click **Next** to configure **Rules**.
   ii. Click **Save Program** to save the program.

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.



2. Click **Save Program** to complete configuration of the program.
3. Click **Finish** to exit the wizard.
4. Configure multicast server and multicast client programs and load all codecs with the appropriate program. [Select and connect audio streams] in a program using the **Connections panel**, or [dial the program manually] using the codec front panel. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast streams.

19.19 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 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: 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. **ETH1** or **ETH2**.
3. Create a SIP program using the front panel via the **Home screen**, or create a SIP program using 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.
- 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 is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless 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 select **Home screen > Settings > 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]**

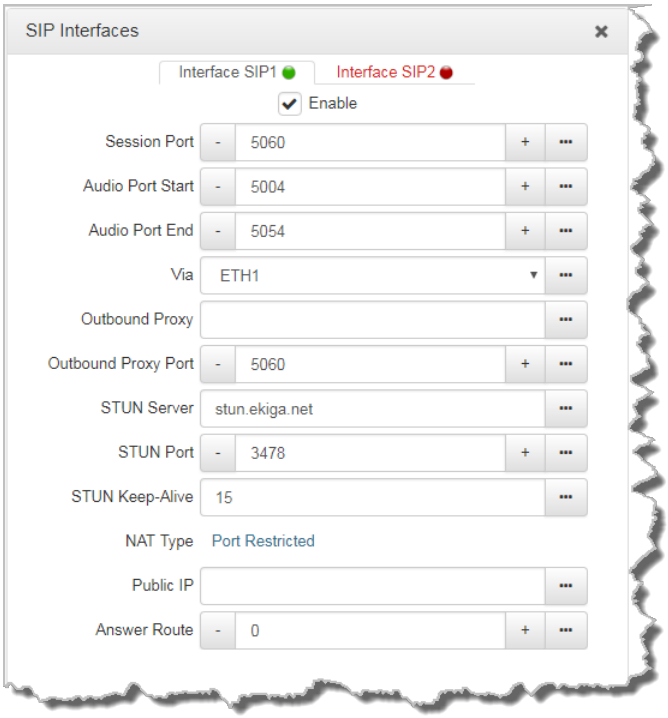
19.19.1 Configuring SIP Interfaces

The codec supports dialing over two SIP interfaces simultaneously.

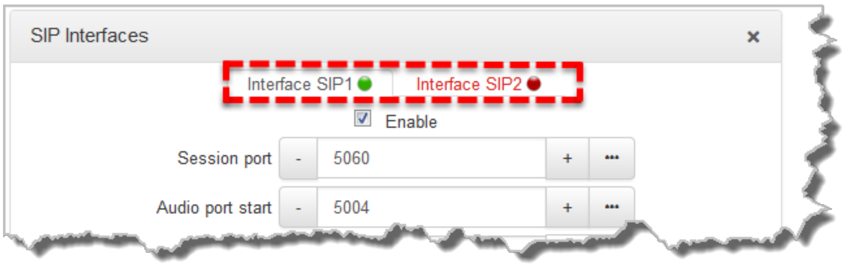
Important Notes:
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **ETH1** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configured to use **ETH2** by default, which is mapped to the **Secondary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **ETH1** or **ETH2**.
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 interface uses a unique "Via" IP interface because they can't share one, e.g. **ETH1**.



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.



19.19.2 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).

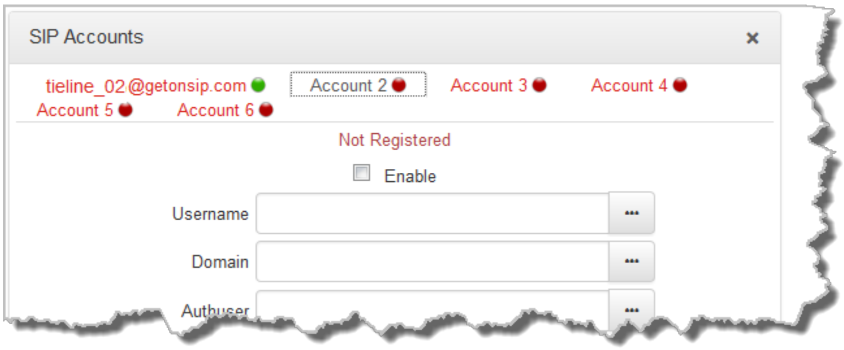
Important Notes:

- 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 **LCD SCREEN** see Configuring SIP Accounts.

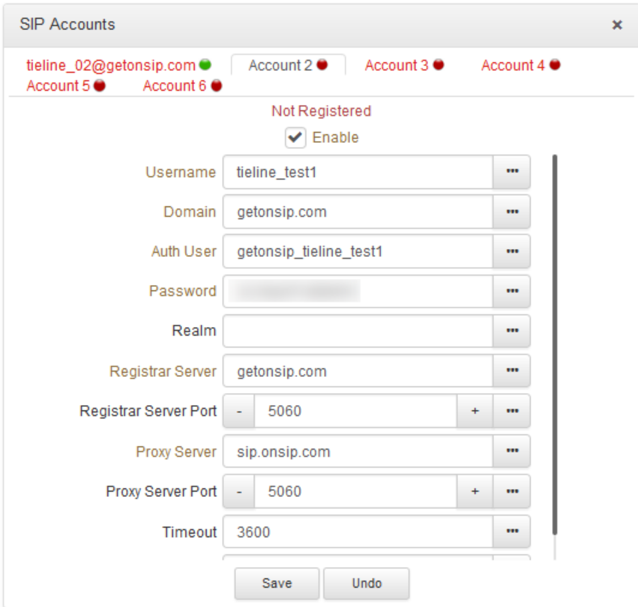
Adding a SIP Account

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.

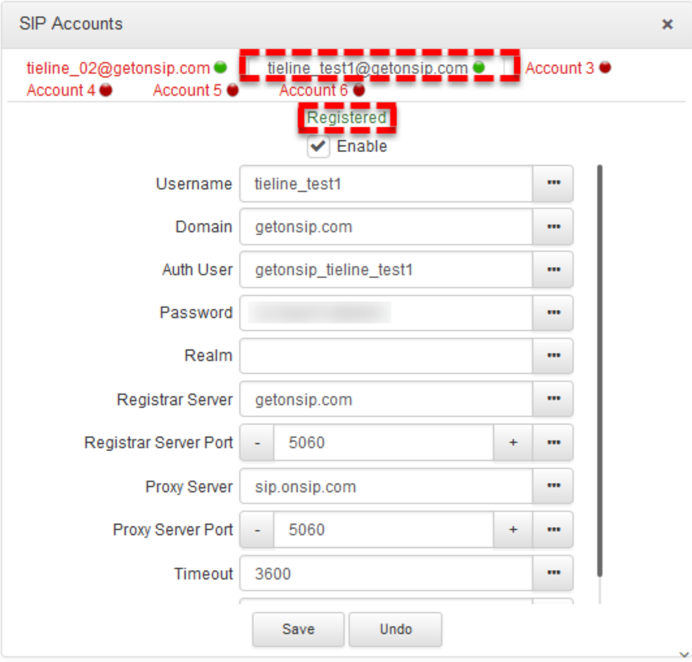
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Accounts** to view and configure SIP account settings.
2. Click to select one of the unused **Accounts** at the top of the **SIP Accounts panel**.



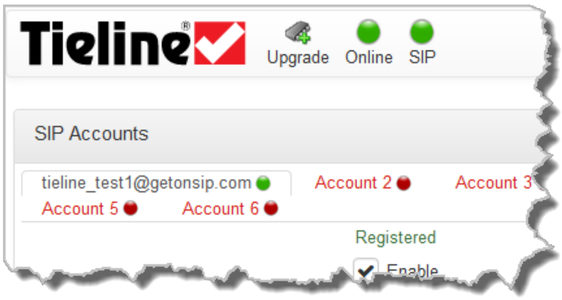
3. Enter the SIP account details into the relevant text boxes, including the registration **Timeout** (which shouldn't need to be adjusted from the default setting). Also ensure a SIP **Interface** is selected (e.g. **SIP1** or **SIP2**.) The SIP interface contains settings related to ports and the selected **Via** interface, e.g. **ETH1** or **ETH2**. 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.

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 as the **Via** in the account is enabled.
3. Verify that the **Via** selection in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **ETH1**, **ETH2**.

19.19.3 Configure SIP White and Blacklists

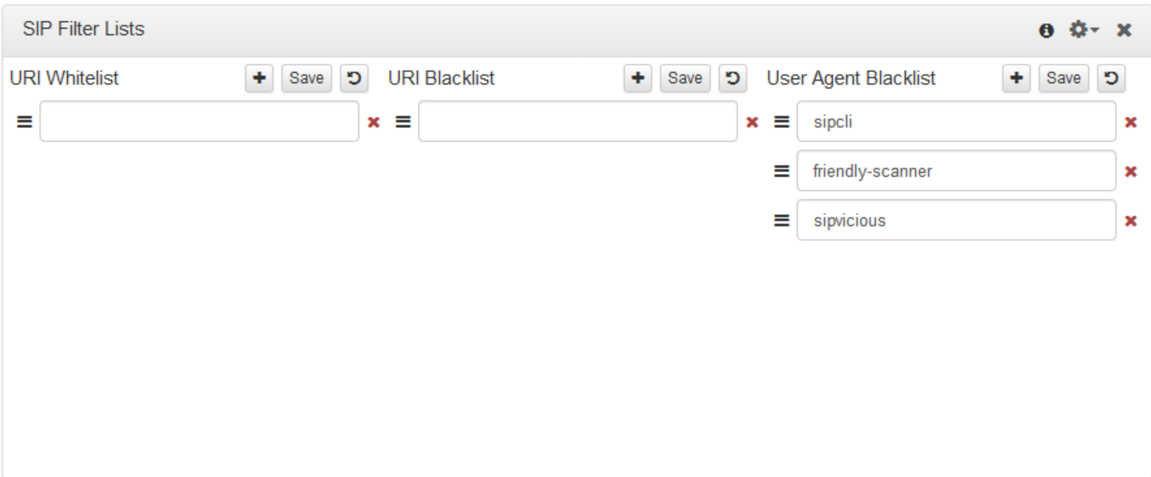
The **SIP Filter Lists panel** allows filtering of SIP URIs and User Agents to provide greater security for your codec connections. For example, add trusted network codecs to the **URI Whitelist** in this panel and only codecs using these SIP URIs will be able to connect. This is like saying, "if you 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 Blacklist** and add user agents to the **User Agent Blacklist** to deny them access to the codec. These blacklists also filter unwanted traffic and increase the likelihood of rejecting unwanted traffic. Note: If an incoming SIP caller is not on the URI Whitelist it will be scanned using the **URI Blacklist**. If there is no match it will be scanned using the **User Agent Blacklist**. A connection will be established if there is no match on either Blacklist.

Important Note: To only allow a predefined list of codecs to connect, add them to the **URI Whitelist** and add a wildcard (asterisk) * to the **URI Blacklist**: all incoming calls will be blocked except for codecs in the Whitelist.

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**.

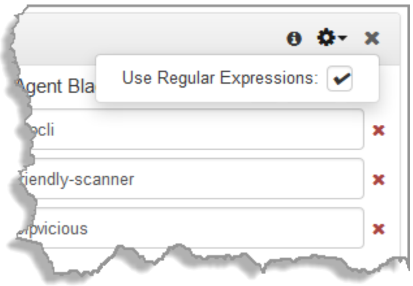


2. Click the **Plus symbol** for **URI Whitelist**, **URI Blacklist** or **User Agent Blacklist** to add a new item to the list.
3. Enter the new item in the text box, click to select the check-box and then click **Save** to store the new setting.
4. Click the **Undo symbol** to undo editing and click and drag the **List symbol** ≡ to shift the position of whitelist and blacklist items.

Important Note: Some codec manufacturers allow whitelisting of calls by 'User Agent.' Therefore, it may be necessary to enter a Tieline codec user agent into a non-Tieline codec to allow it 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

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.



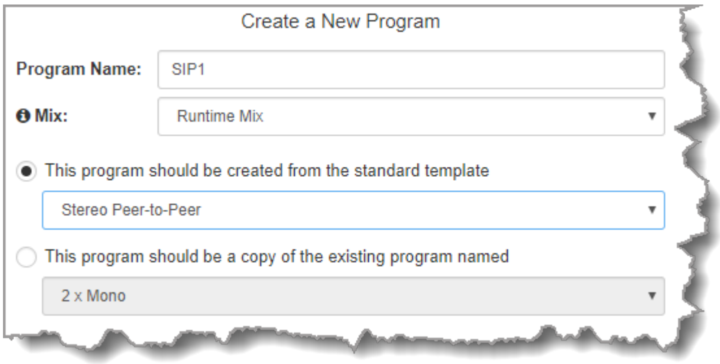
Important Note: Regular expressions should not use ^ and $ anchors because searches implicitly try to match anywhere in the line.

19.20 Configure Peer-to-Peer SIP Programs

SIP programs are like a normal IP program to configure, with two small differences; entering a SIP address and selecting SIP as the **Session Protocol**.

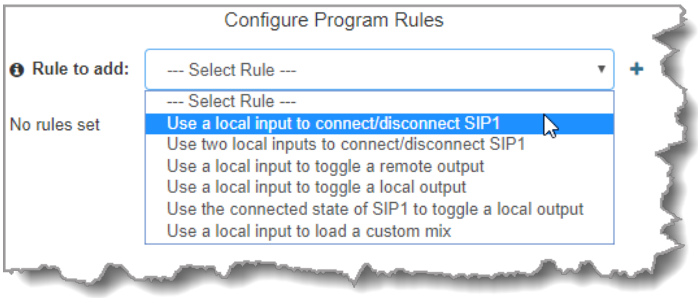
Important Notes: Before you start program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- To learn more about programs see the section titled About Program Dialing.
- 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.

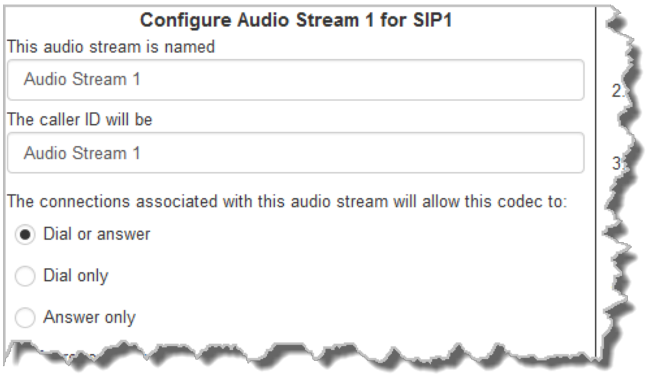


Important Notes for Rules:

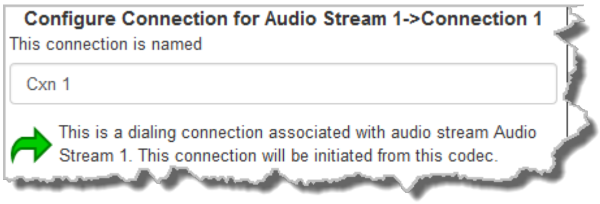
- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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 will display 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, TieLink, 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.



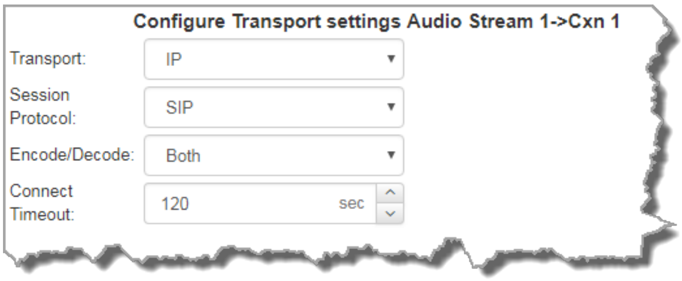
5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



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**.

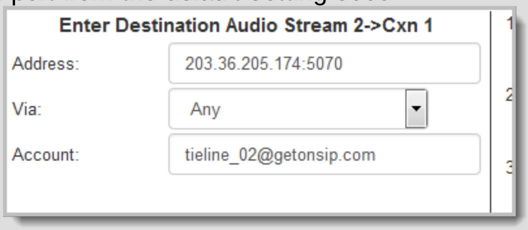


7. Configure the destination codec **Address** if you are dialing peer-to-peer, then specify the network interface used to dial the connection, e.g. **Primary** (Ethernet port 1). Enter the name of a registered SIP account if you are using a SIP server to establish a connection. If you wish to dial from one of the codec's registered accounts, then enter the account name in the Account field using the format accountname@sipserverdomain, e.g. tieline_test1@getonsip.com. In this configuration the account interface will be used rather than the specified **Via**, e.g. if the account is using **SIP2** and this is configured to use Ethernet 2, then the call will proceed using Ethernet 2. If you do not wish to use an account for the dial then leave the **Account** field blank and select the required interface. Note: the interface must be associated with either **SIP1** or **SIP2** for the call to be able to proceed.

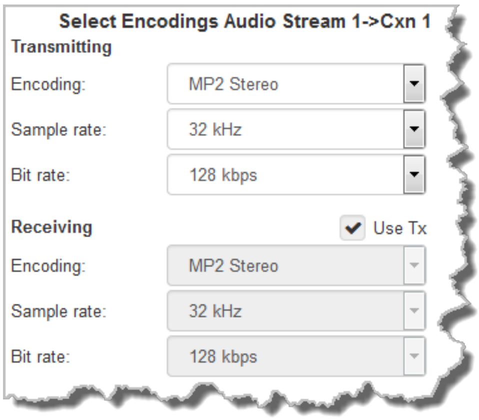
At this point you can click **Save Program** and save the program with default algorithm and jitter settings. Alternatively, click **Next** to confirm and specify algorithm and jitter settings for this connection and configure backup audio settings (recommended).

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.

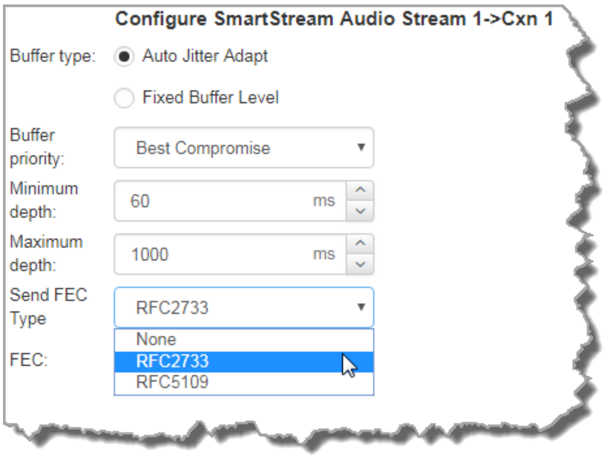


8. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click **Next** to continue.



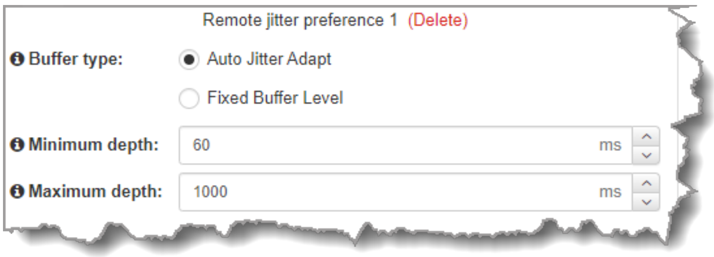
9. Click to configure:

- **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.

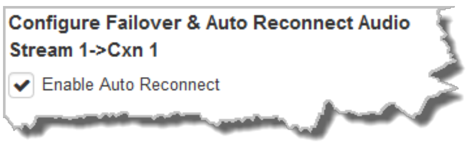


10. Click **Add a remote jitter preference** to send preferred jitter settings to a remote codec. Note: this is just a preference as per EBU Tech 3368 and there is no guarantee that the remote codec will accept or support these jitter configuration settings. Verify configuration settings on the remote codec to ensure settings are correct. Recommended jitter buffer limits are as follows:

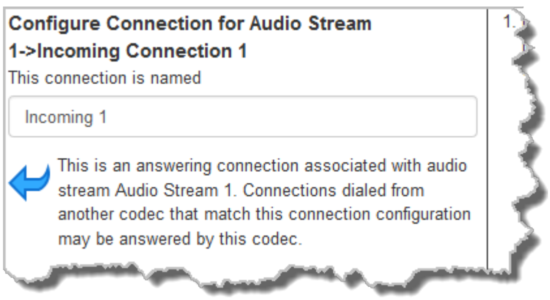
- 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
- 2,500ms for AAC ELD, AAC LD.
- 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS.



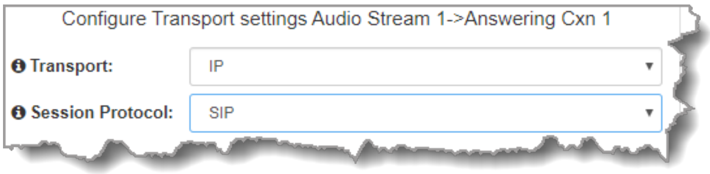
11. Click **Next** to select the check-box if you want to **Enable Auto Reconnect**.



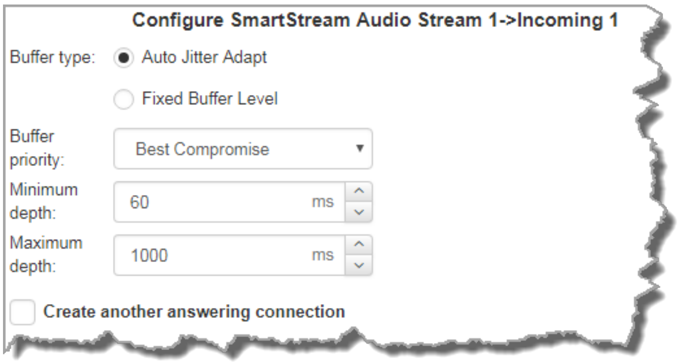
12. Click **Next** to name the answering connection for when calls are received by the codec.



13. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.



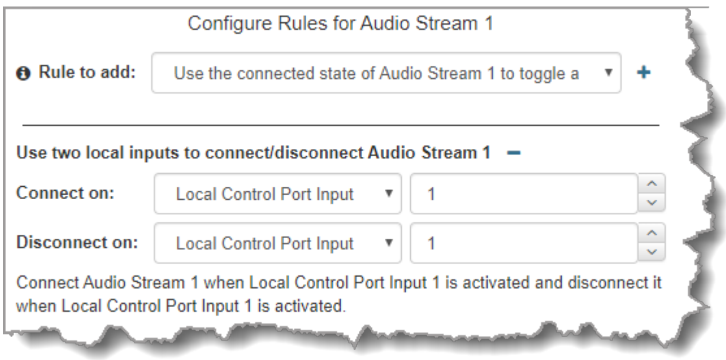
14. Click **Next** to configure the jitter buffer settings for the answering connection.



14. 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 Program** to save the program at this point.
   iii. Click **Next** to configure rules options.

15. 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.

16. Click **Save Program** to save the program.
17. Click **Finish** to exit the wizard.
18. 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.

19.21 Configure Multiple Stream SIP Programs

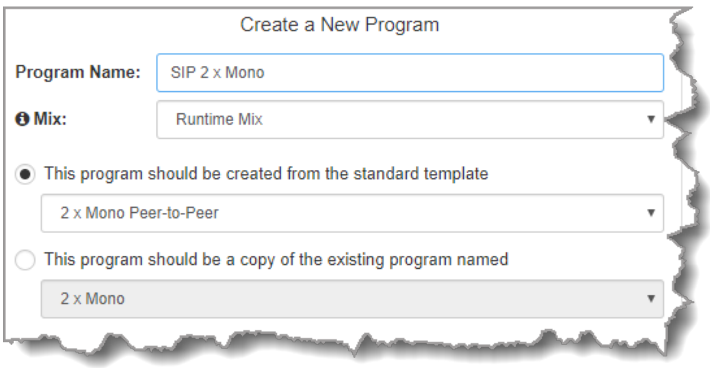
To configure a multiple stream SIP program simply create a new program and configure each SIP audio stream as you would for a single SIP Peer-to-Peer program. In the following example a **2 x Mono Peer-to-Peer** dial and answer program is being configured.

Configuring a 2 x Mono Peer-to-Peer SIP Program

Important Notes: Before you start program configuration please note:

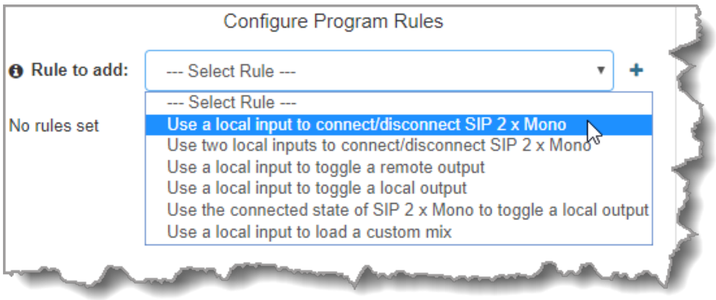
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- To learn more about programs see the section titled About Program Dialing.
- 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 **2 x Mono 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.

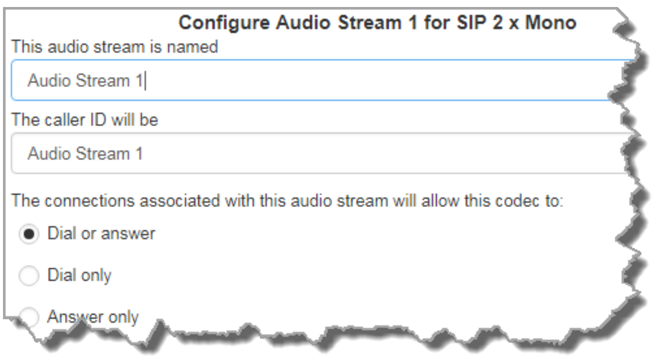


Important Notes for Rules:

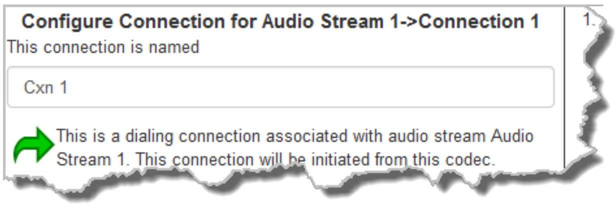
- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- 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 will display 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. When answering multiple SIP connections, answer routes can be used to create deterministic dialing, which routes audio from incoming calls consistently to the same inputs and outputs. For more information please see Answering Multiple SIP Peer-to-Peer Programs. Please note that caller ID, dial routes, TieLink, 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.



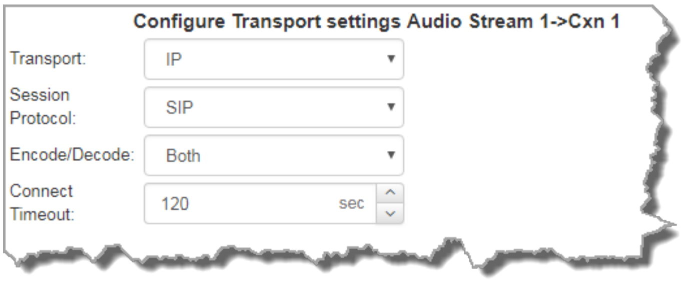
5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



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**.

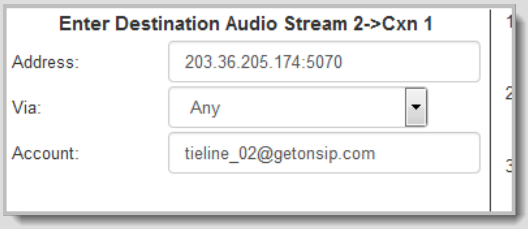


7. Configure the destination codec **Address** if you are dialing peer-to-peer, then specify the network interface used to dial the connection, e.g. **Primary** (Ethernet port 1). Enter the name of a registered SIP account if you are using a SIP server to establish a connection. Use the format accountname@sipserverdomain, e.g. tieline_test1@getonsip.com

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).

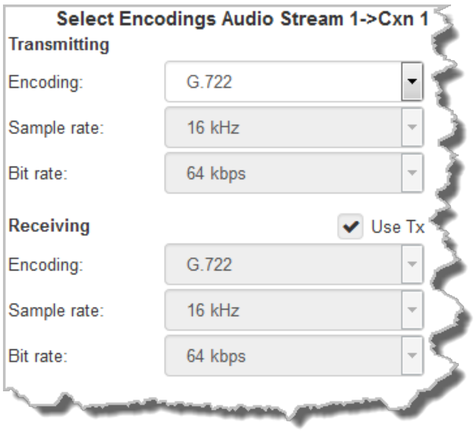
Important Notes:

- 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.



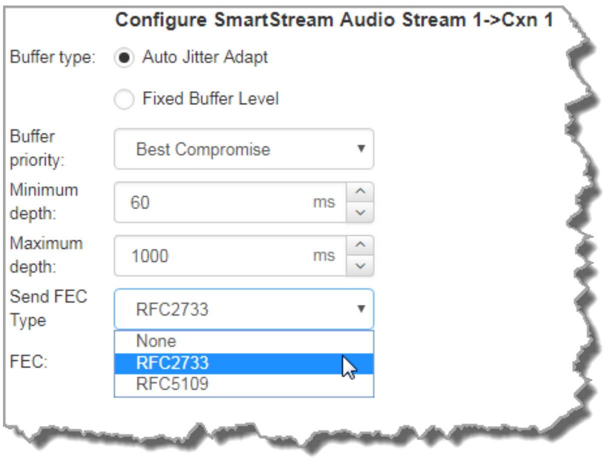
- The audio port used by the codecs is allocated automatically from the 'pool' of ports specified in the **SIP Interfaces** panel. The default setting is 5004 to 5054.
- To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open 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.

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.



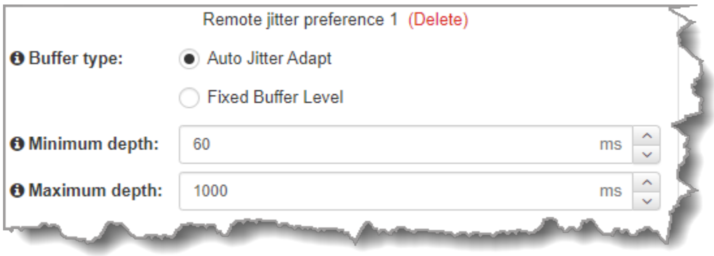
9. Click to configure:

- **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.

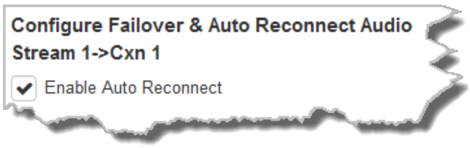


10. Click **Add a remote jitter preference** to send preferred jitter settings to a remote codec. Note: this is just a preference as per EBU Tech 3368 and there is no guarantee that the remote codec will accept or support these jitter configuration settings. Verify configuration settings on the remote codec to ensure settings are correct. Recommended jitter buffer limits are as follows:

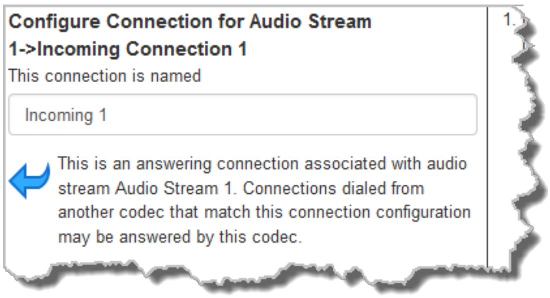
- 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
- 2,500ms for AAC ELD, AAC LD.
- 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS.



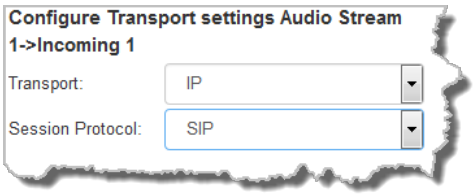
11. Click **Next** to select the check-box if you want to **Enable Auto Reconnect**.



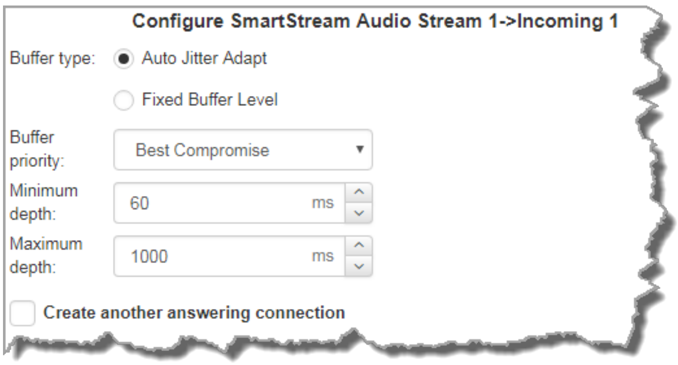
12. Click **Next** to name the answering connection for when calls are received by the codec.



13. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.



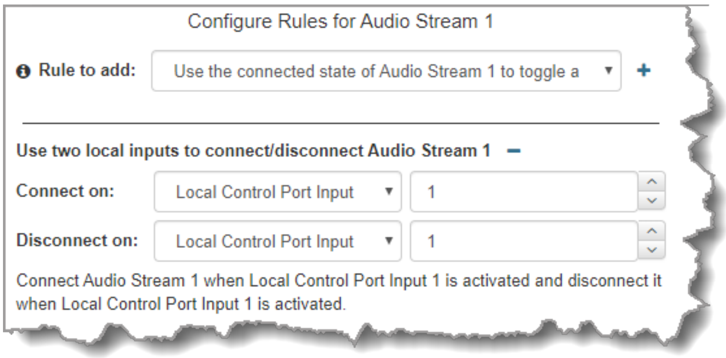
14. Click **Next** to configure the jitter and FEC settings for the answering connection.



15. 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 **Next Stream** to configure the next audio stream.
    iii. Click **Next** to configure rules options.

16. 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.

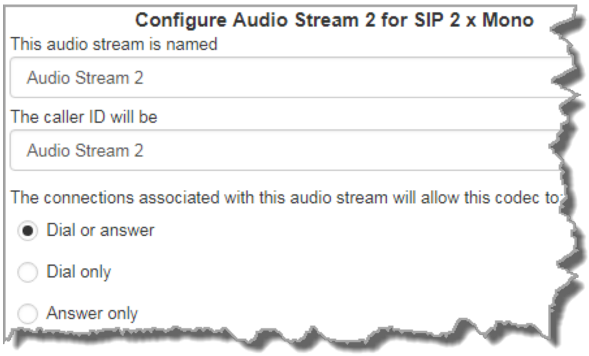


Important Notes for Rules: Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

Configure the Second Peer-to-Peer SIP Audio Stream

1. Click **Next Stream** to enter the second **Stream Name** and configure the codec to dial, answer or dial and answer. Then click **Next**.

Note: When answering multiple SIP connections, answer routes can be used to create deterministic dialing, which routes audio from incoming calls consistently to the same inputs and outputs. For more information please see Answering Multiple SIP Peer-to-Peer Programs. Please note that caller ID, dial routes, TieLink, 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.



2. Continue through the wizard and configure the second dial and answer stream connections in a similar way to the first audio stream connections. Then click **Save Program** to save all settings and click **Finish** to exit the Program Manager wizard. 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.

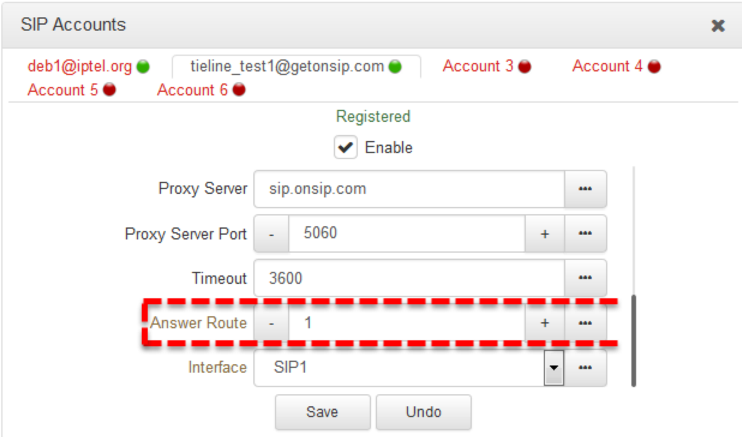
Important Note: Remember to lock an answering program in a codec when answering multiple SIP calls.

19.22 Answering Multiple SIP Peer-to-Peer Calls

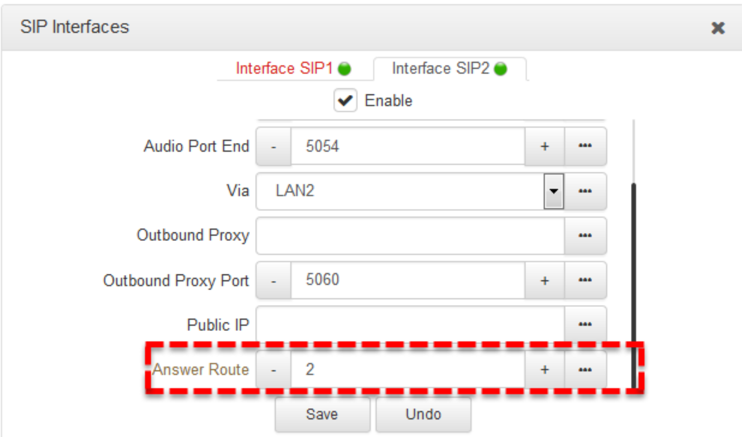
The codec is capable of accepting multiple SIP Peer-to-Peer connections and routing them in a deterministic manner to specific audio input and outputs.

Tieline session data, in conjunction with unique audio ports, normally facilitates routing of multiple incoming streams to specific inputs and outputs on a single answering codec. Session Description Protocol (SDP) is used over SIP instead of Tieline session data, therefore deterministic routing of incoming calls needs to be done a bit differently. There are two options:

1. **SIP Accounts**: Using the **SIP Accounts panel** in the HTML5 Toolbox web-GUI, register up to 6 SIP accounts and configure a unique answer **Route** in each SIP account. Then create a multiple stream answering program (e.g. **2 x Mono Peer-to-Peer**) and configure an answer route in each audio stream matching the routes used for each SIP account you have registered to the codec. In this way, when a call is received via a particular account, the call is routed to the audio stream with the matching audio answer route. This process also reliably predetermines which inputs and outputs are used on the answering codec (e.g. **Input 1** and **Output 1** for the first audio stream in a multiple stream program).



2. **SIP Interfaces**: When answering multiple incoming peer-to-peer SIP calls, configure an answer **Route** for incoming peer-to-peer calls via the **SIP Interface panel** and route them to specific audio streams. Configure answer routes in **Interface SIP1** and **Interface SIP2** to match answer routes configured in an answering program's audio streams. This process also reliably predetermines which inputs and outputs are used on the answering codec (e.g. **Input 2** and **Output 2** for the second audio stream in a multiple stream program).



Important Notes:

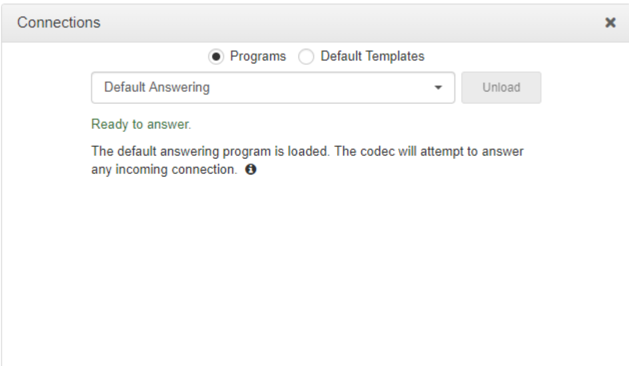
- Ports are arbitrarily assigned when answering calls using SIP accounts. By default the UDP audio port range used is from 5004 to 5054. This setting can be adjusted in the **SIP Interfaces panel** in the HTML5 Toolbox web-GUI. Ensure your firewall has the required TCP and UDP ports open if you are receiving multiple SIP calls.
- 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 multiple SIP audio streams to connect. See Installing the Codec at the Studio for more information.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.

19.23 Load, Unload and Dial a Program

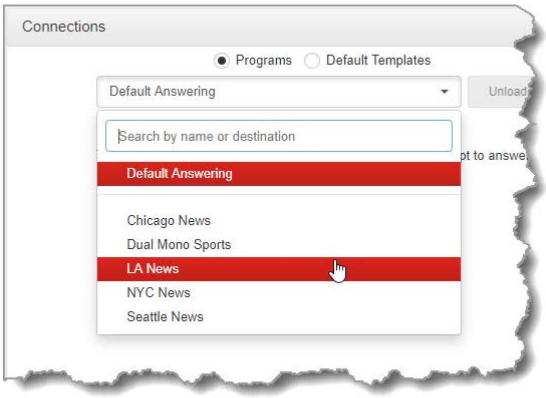
A program can be loaded, unloaded and edited using the **Program Manager panel** or the **Connections panel**. Audio stream dialing settings can be edited without unloading a program, even if other audio streams are concurrently connected. Please note: If you need to use an existing program as a template for a new program, or add a custom matrix mix to a program, you must use the **Program Manager panel** to create a program.

Load and Unload a Program

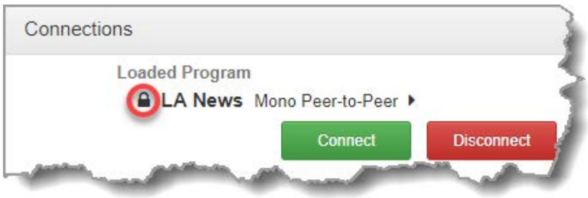
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**.



2. Select **Programs** and then click the drop-down arrow to select one of the programs in the codec, then click **Load**.



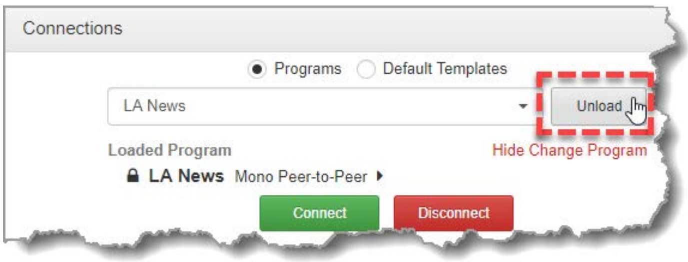
3. If **Lock Loaded User Program** has been configured in the **Options panel**, a black **Padlock** symbol appears next to the program name in the **Connections panel**, to indicate a program is locked in the codec.



Unload a Program

1. To unload a loaded program click **Change Program**.

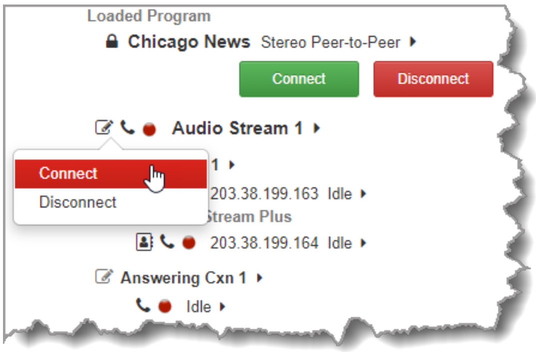
2. Click **Unload**.



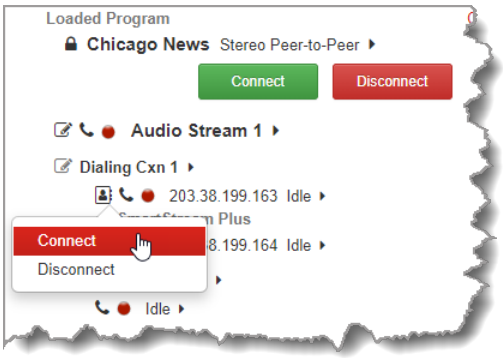
Connecting a Program

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 **Connect/Disconnect** symbol and then click **Connect**; this connects all connections associated with this audio stream.

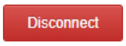


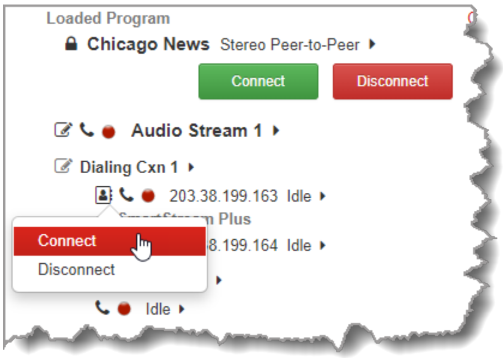
3. Click the connection **Connect/Disconnect** symbol and then click **Connect**; this connects an individual audio stream connection.



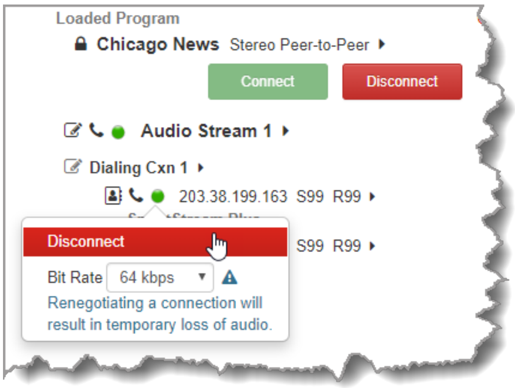
Disconnecting a Program

To disconnect audio streams and connections within an existing program there are three options:

1. Click the **Disconnect**  button to disconnect all audio streams and connections configured in a program.
2. Click the audio stream **Connect/Disconnect** symbol and then click **Disconnect** to disconnect an individual audio stream and all associated connections.



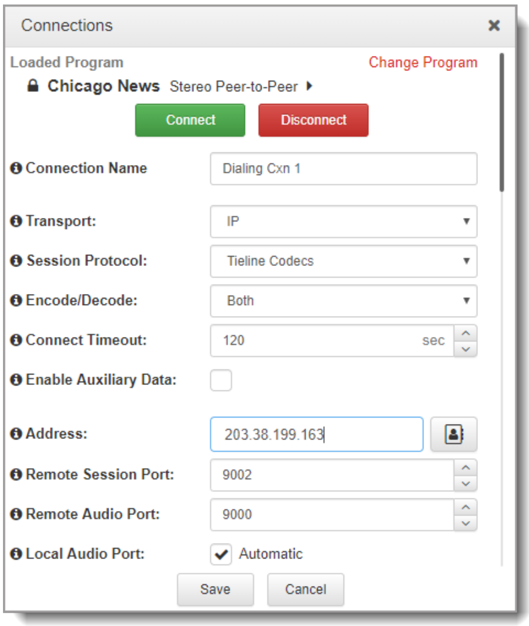
3. Click the connection **Connect/Disconnect** symbol to disconnect an individual audio stream connection.



Change Dialing Settings

To edit destination dialing settings:

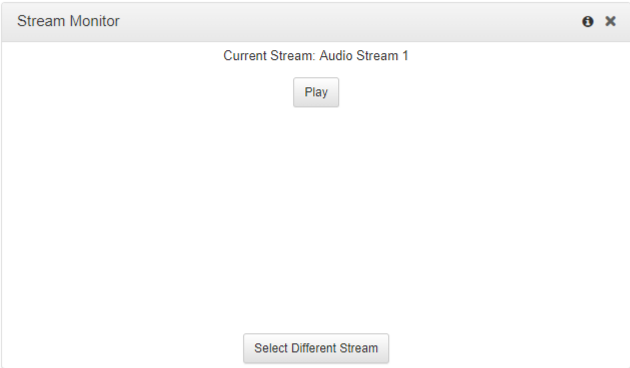
1. Click the **Edit symbol** adjacent to a connection.
2. Adjust dialing and connection settings and then click **Save** to change edited settings in the program. Note: The IP address can be changed, or a **TieLink contact** can be selected if Traversal Server Contact Lists have been configured. See Configuring TieLink Settings for more info.



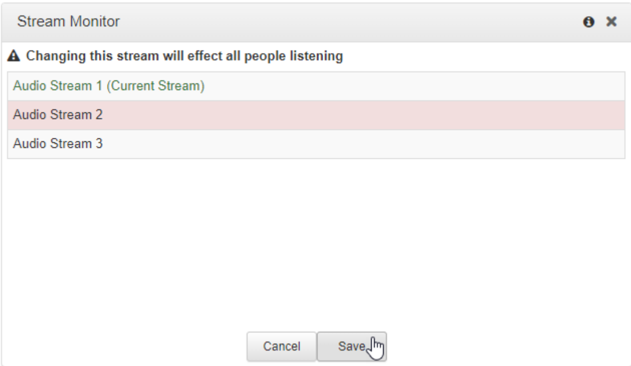
19.24 Monitor Stream Audio

It is possible to select and monitor the audio from individual audio streams using the Toolbox HTML5 Web-GUI.

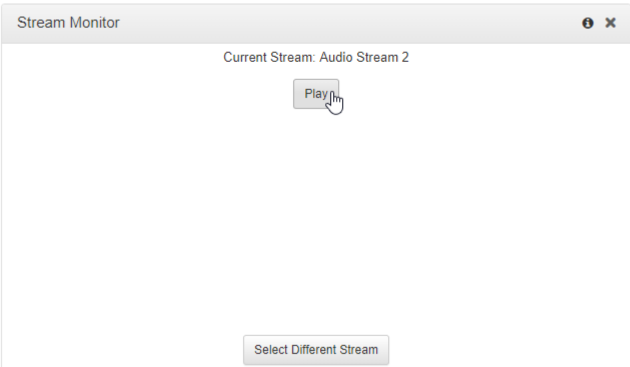
1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Stream Monitor** to display the **Stream Monitor panel**.



2. Click **Select Different Stream** and then click to highlight a new stream to monitor. Next, click **Save**.



3. Click **Play** to monitor audio from the newly selected stream.



Important Note: Audio in the **Stream Monitor panel** will be delayed a few seconds or more, depending on the browser used and data capabilities of the IP network.

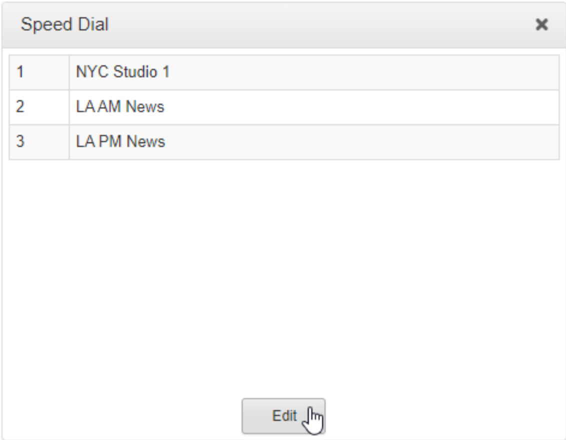
19.25 Configure Speed Dialing

The Toolbox HTML5 web-GUI supports dialing programs using preconfigured speed dials. This is configured and managed using the **Speed Dial panel**.

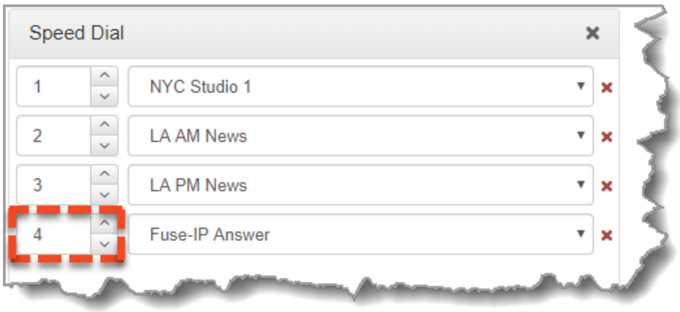
Create a New Speed Dial

To configure speed dialing:

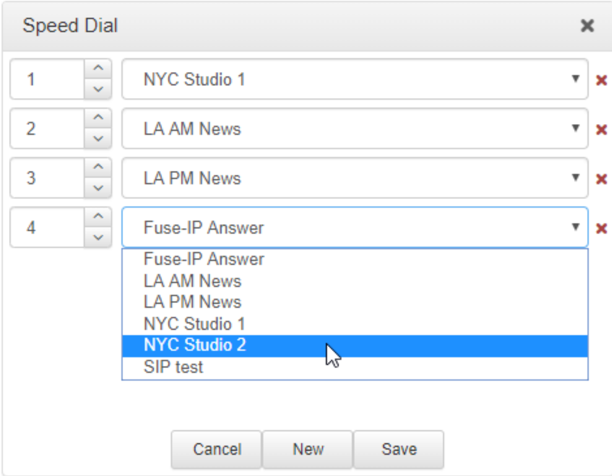
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Speed Dial** to open the **Speed Dial panel**.
2. Click **Edit**.



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.

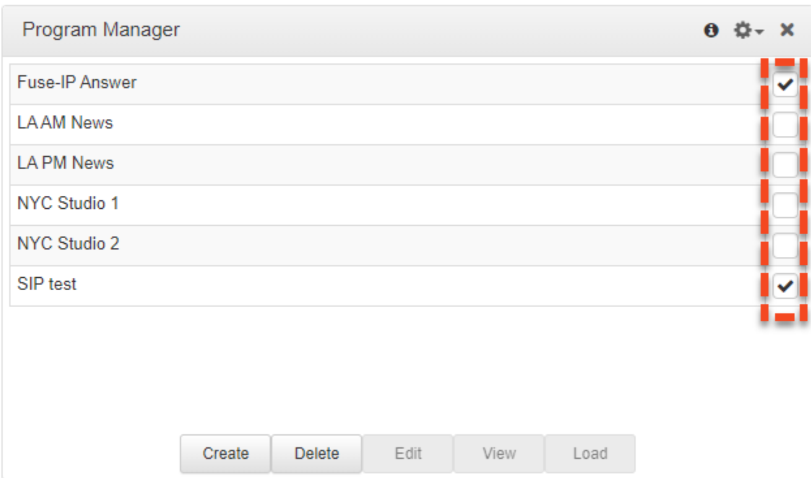


5. Click **Save** to store all edited speed dials.

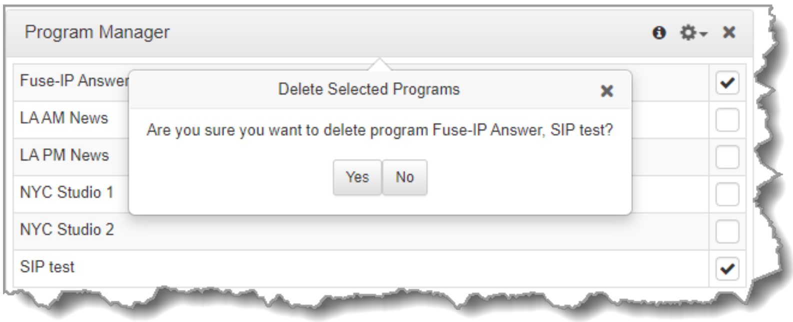
Important Note: To view speed dial configurations using the front panel of the codec select **SETTINGS** **> Speed Dial** and press the button.

19.26 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.

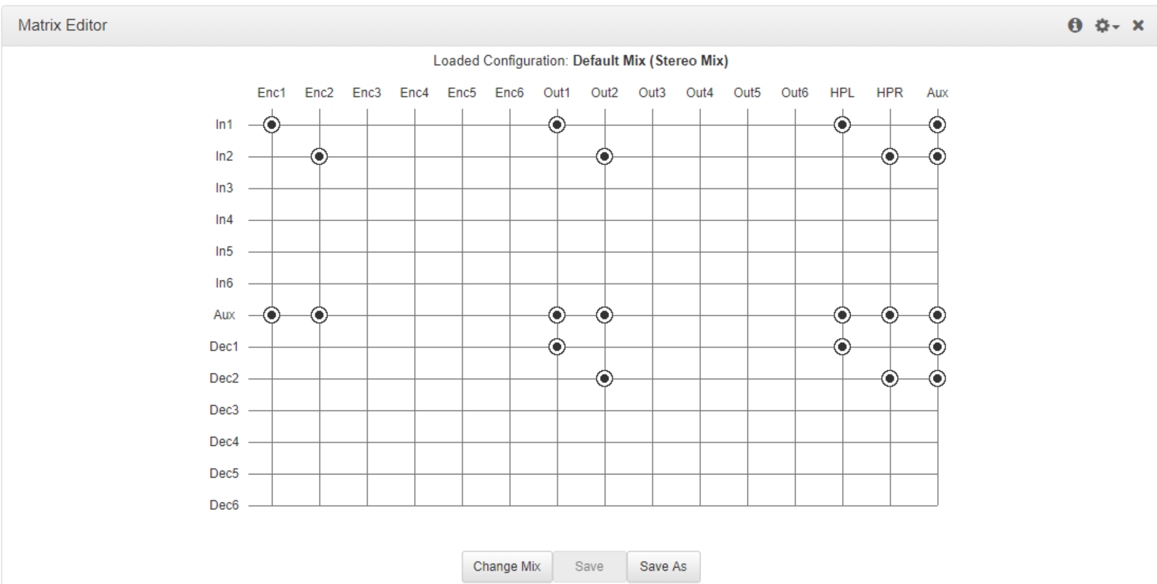
19.27 Matrix Editing

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

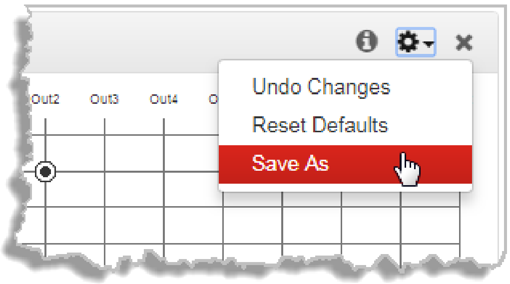
Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Matrix Editor** to open the **Matrix Editor panel**. In the following image the default stereo matrix is loaded, as indicated at the top of the panel.



Click the **Options symbol** ⚙ to view other matrix editor options, including:

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.

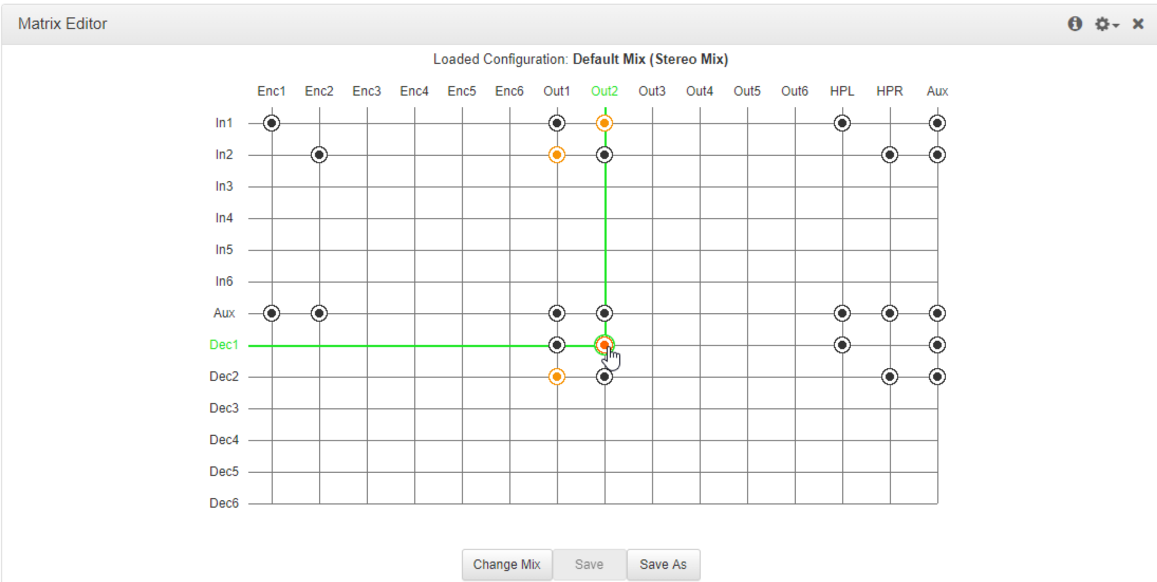
3. **Save as**: Save the current matrix settings as a new **Custom Mix** with a unique name (includes headphone and Cue/TB matrices).



Important Notes: Click the **Information symbol** at the top of the panel to view details about the current matrix.

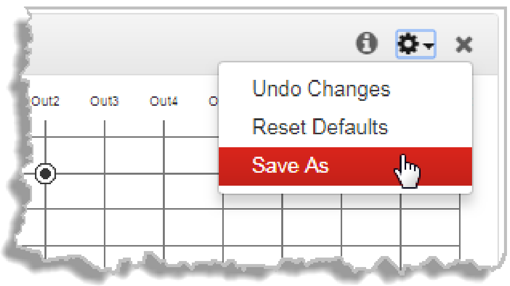
Using the Matrix Editor

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.

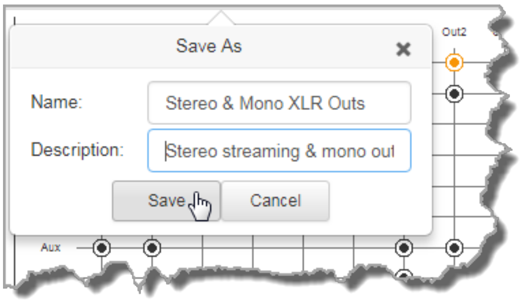


Edits can be saved as a custom mix as follows:

1. Click the **Options symbol** and select **Save As**.



2. Enter a new **Name** and **Description** (if required) and click **Save**.



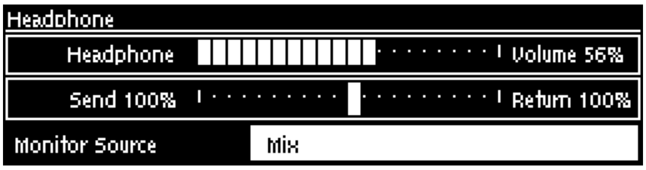
3. The new custom mix stays loaded in the codec until a new mix is loaded, an incompatible program is loaded, or program defaults are restored via **Reset Defaults**.

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.

Headphone Monitoring

To monitor matrix editor audio routing from either headphone output select **HEADPHONE** **> Monitor Source > Mix**. This setting monitors runtime audio changes in real-time.

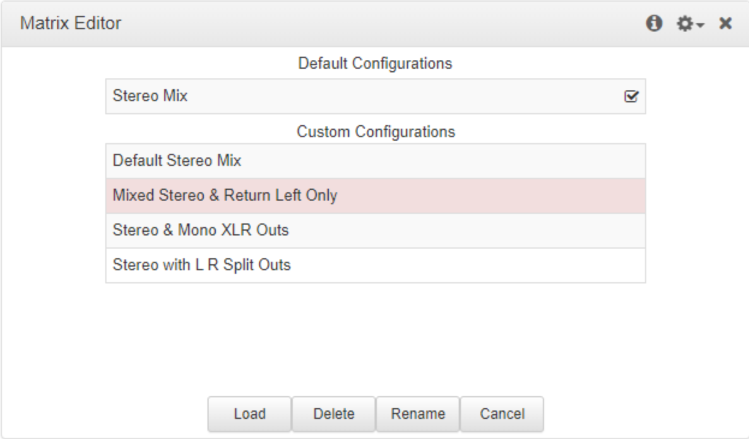


Important Note: Codec front panel **PPMs 1-6** and PPMs in the **PPMs panel** always reflect the default program settings and do not display customized mix adjustments using the **Matrix Editor**.

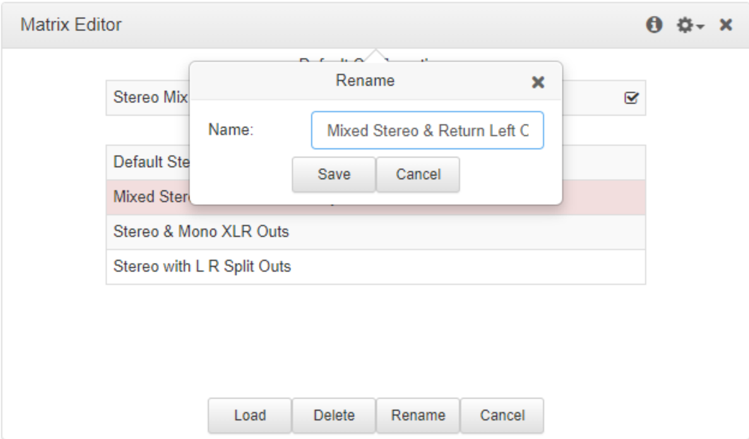
Load, Rename and Delete a Custom Matrix Editor Mix

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.

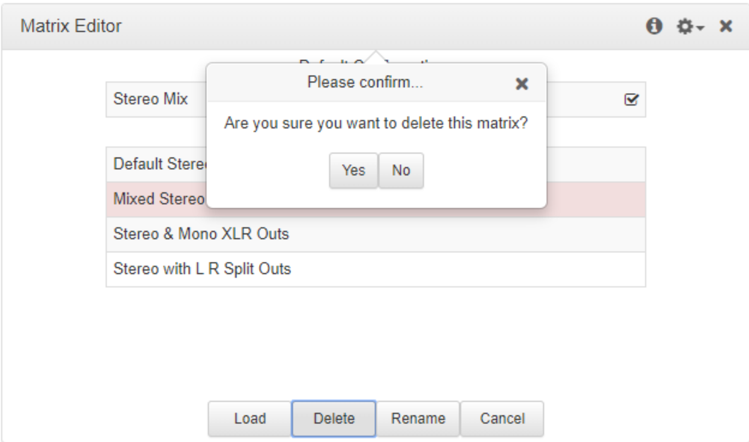
1. To load a custom mix click **Change Mix**.



2. To rename a custom mix, click to select a saved mix and then click **Rename**. Edit the name and click **Save**.

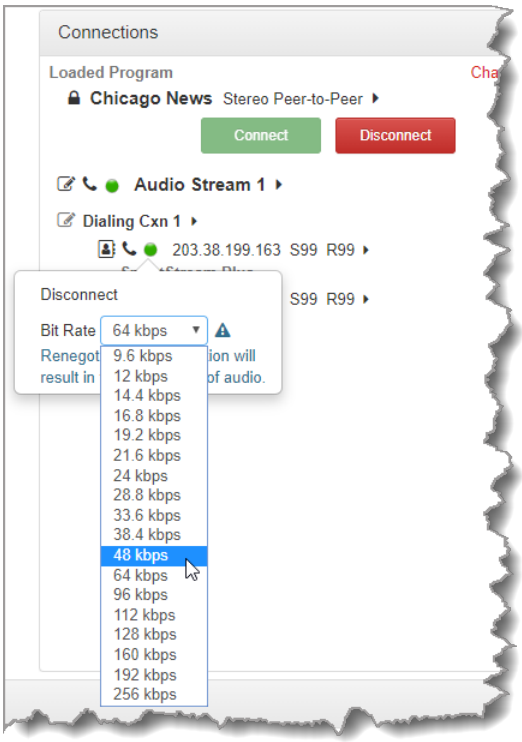


3. To delete a custom mix, click to select a custom mix and then click **Delete**. Note: A loaded mix cannot be deleted.



19.28 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 **Connect/Disconnect** symbol for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit-rate.

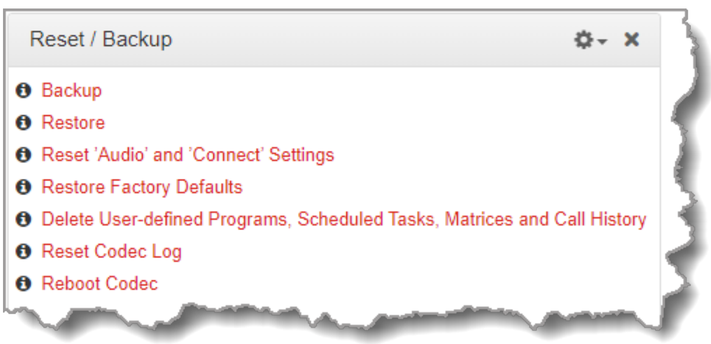


Important Notes:
- Renegotiation will result in a temporary loss of audio.
- It is not possible to renegotiate the connection bit rate of a SIP connection.

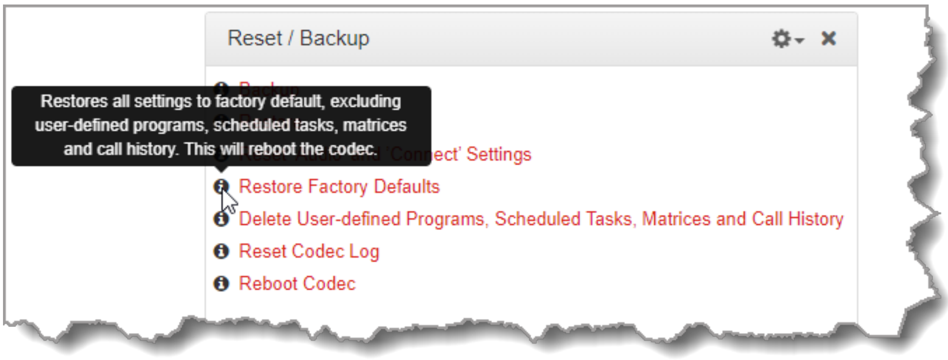
19.29 Reset Factory Default Settings

There are several options which allow you to restore default settings within the codec. See Reset and Restore Factory Defaults for more details on each option.

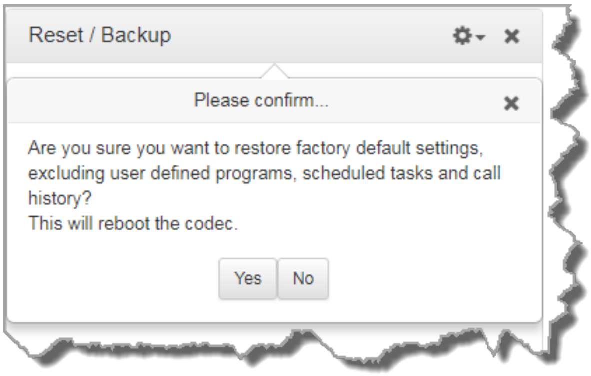
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset / Backup** to display the **Reset / Backup panel**.



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; click **Yes** to proceed.



19.30 Backup and Restore Functions

The HTML5 Toolbox Web-GUI can be used to backup and restore codec settings, including:

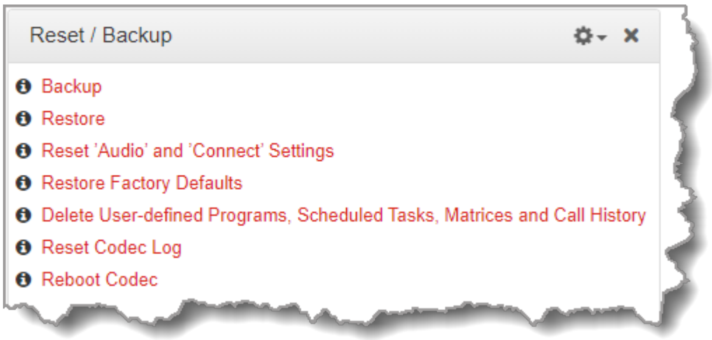
- Programs containing a variety of connection settings and scheduler configuration settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

Files can also be used to copy configurations onto other similar codecs. Programs are essentially connection profiles that may include:

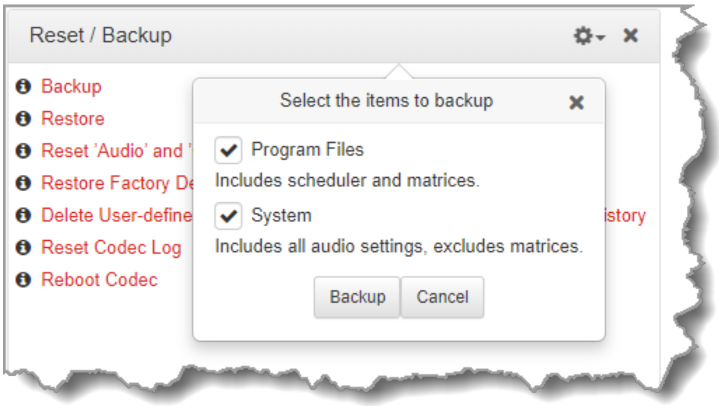
- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

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**.



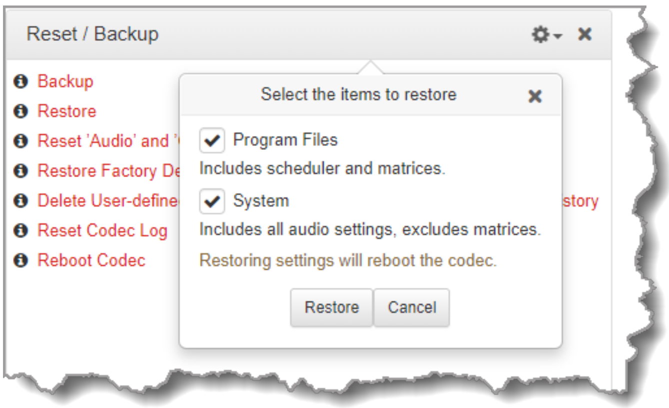
2. Click **Backup**.
3. Click to select the check-boxes to confirm your backup requirements, then click **Backup**.



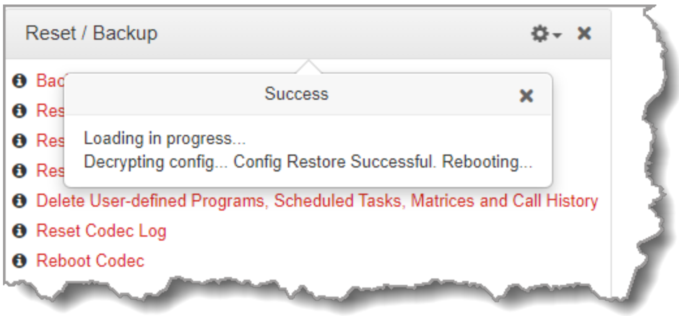
4. Select a location on your PC to save the configuration file. Note: You may need to "allow" your browser to display the pop-up dialog.

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 to only copy programs onto codecs.



4. Click **Restore** and select the .tgz file you want to load onto the codec. A **Success** dialog confirms the files have been restored.



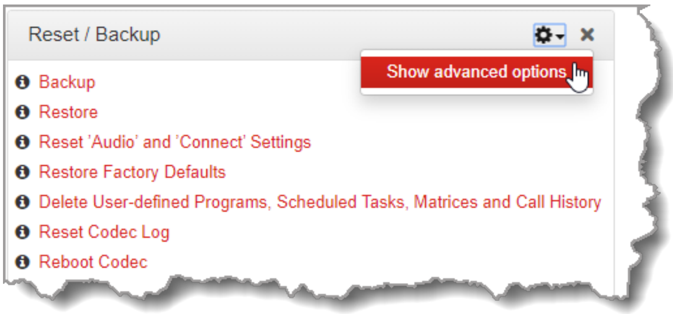
Note: The codec will automatically reboot when restoring system settings.

Advanced Settings: XML Config

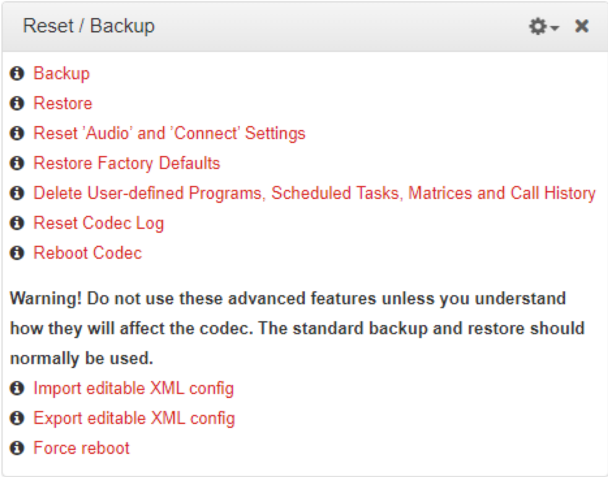
Caution: DO NOT use advanced XML configuration features unless you fully understand how they will affect the codec. The standard backup and restore function should normally be used. Damage to the codec may occur if this feature is used without fully understanding how it will affect the codec.

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.

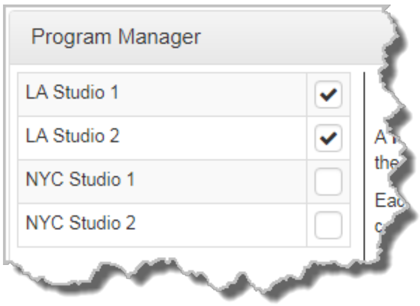


19.31 Import and Export Programs

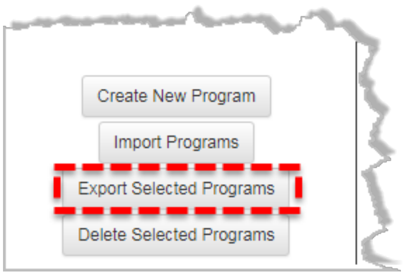
It is possible to import and export individual programs using the **Program Manager panel**.

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 program or programs you wish to export.



3. Select **Export Selected Programs** in the bottom-left corner of the **Program Manager panel**.

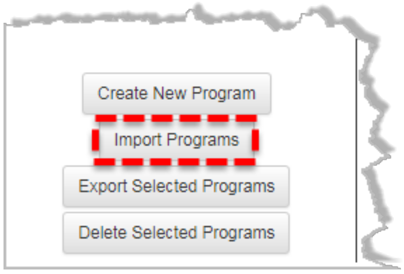


4. Navigate to a file folder and save the program .zip file.
5. Click **Save** to save the program file.

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. Select **Import Programs** in the bottom-left corner of the **Program Manager panel**.

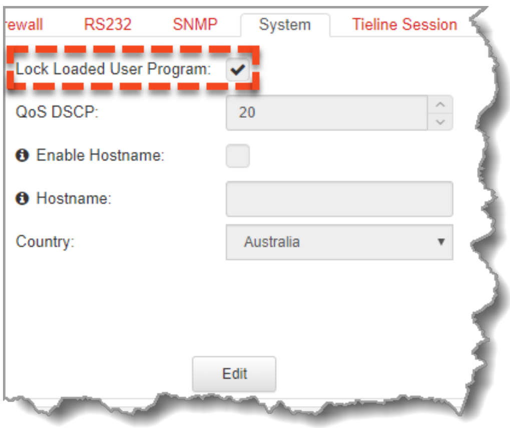


3. Navigate to the file folder containing the program .zip file to be imported.
4. Click to select the .zip file and click **Open** to import it.

19.32 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 a user program 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.

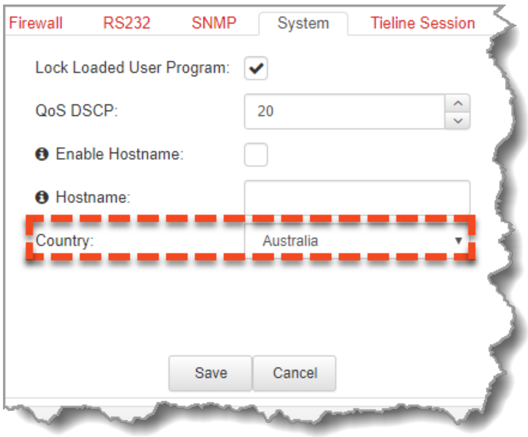
19.33 Configure Country Setting

The **Country** setting in the codec configures country-specific settings like:

- POTS line ring tones.
- POTS line impedance.
- Use of G.711 µ-law for North America and Japan, and G.711 a-law in most other regions of the world (e.g. Europe/Australasia), when the G.711 algorithm is used for IP/SIP or ISDN connections.

To configure the **Country** setting:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **System** and then click **Edit**.
3. Click the **Country** drop-down menu arrow to select the correct country of operation.



4. Click the **Save** button to save the new configuration.

19.34 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:

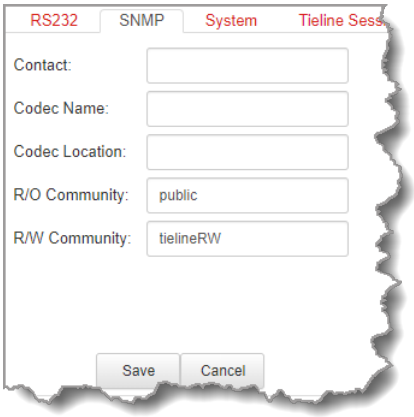
1. Configure SNMP Device settings in your codec.
2. Configure SNMP Traps via the **Alarms Panel** in the Web-GUI (see SNMP Trap Configuration in Configuring Alarms, or to configure using the codec front panel see Configuring SNMP Settings).

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.
Trap Destination	SNMP provides the ability to send traps (notifications or alerts), which are packets containing data relating to a system component. The destination is the end-point to which notifications and alerts are sent. See SNMP Trap Configuration.

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.



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. The required MIB files can be downloaded from the codec using the following link in a PC web browser connected to the same network as your codec:

- http://<YOUR_CODEC_ADDRESS>/mibs/tieline-mibs.zip

Save the .zip file to your PC and import the contents into the MIB browser you use to manage SNMP-enabled network devices.

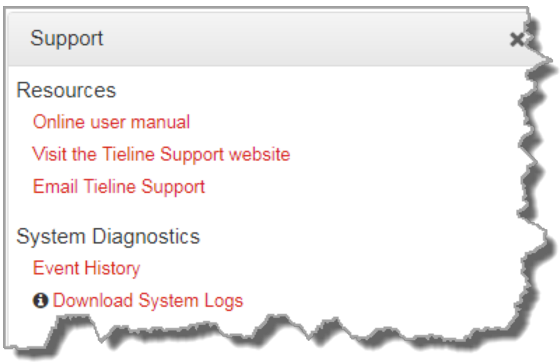
Important Note: The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213. An example of a free MIB browser is available at http://www.ireasoning.com/.

19.35 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.

19.36 Using the Program Scheduler

The program **Scheduler** is a powerful tool which facilitates automatically connecting and disconnecting programs using a simple calendar-based user interface. Key features include:

- Drag and drop to add programs into the scheduler.
- Automatically load and unload programs.
- Drag the top or bottom of a scheduled program to adjust the scheduled time.
- Customization of time-zones displayed in the scheduler.
- Day, week, month, list and timeline views available.
- View a list of scheduled upcoming 'events' in a separate panel.
- Enable and disable scheduled events in a snap.

Scheduler Overview

There are several panels associated with the Program Scheduler:

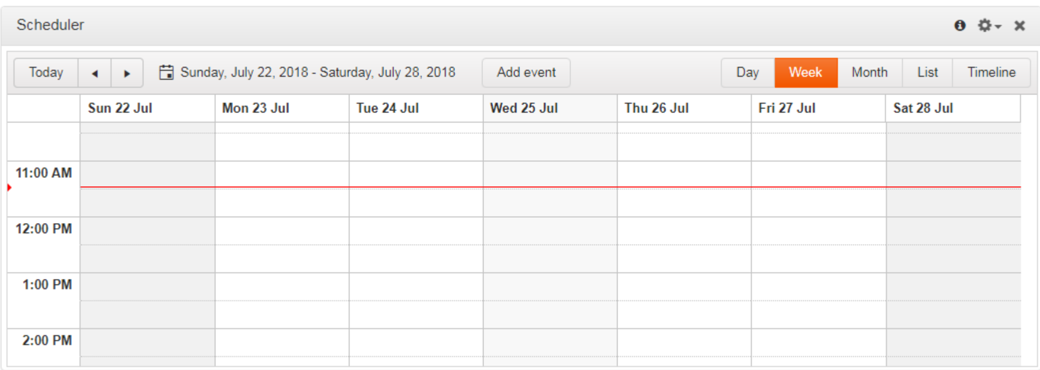
1. **Scheduler**: Create and manage scheduled "events."
2. **Scheduler Events**: View a list of all scheduled events.
3. **Scheduler History**: View previously scheduled events.

Important Notes:
- A program attempts to dial and connect at the scheduled event start time. Allow enough dial time when scheduling events.
- Allow sufficient time to disconnect and dial between consecutive events.
- A scheduled event will fail if a program is already connected.
- Consider configuring events in UTC time during daylight saving transitions to simplify set up.
- Events shorter than 15 minutes display a 15 minute time slot in the scheduler to enhance event visibility.

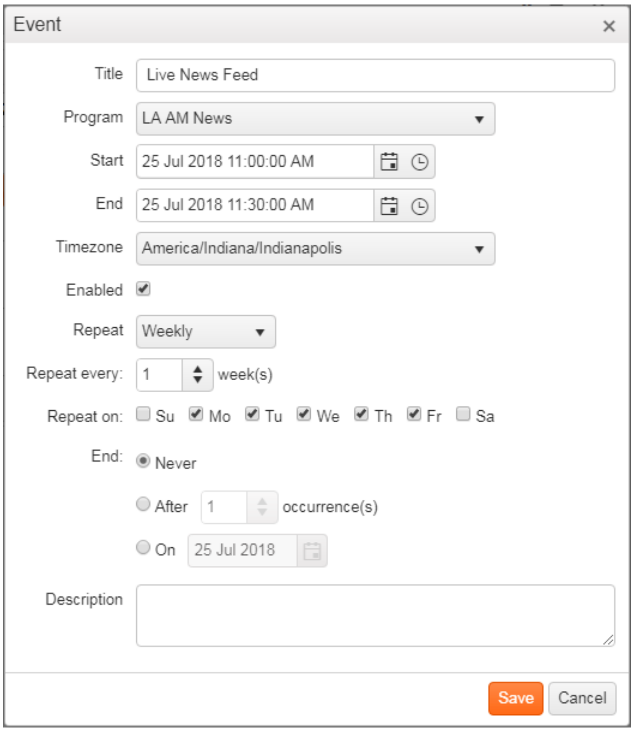
Scheduling New Events

To launch the **Event dialog** and schedule a new program event:

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Scheduler** to launch the **Scheduler panel**.
2. To add a new event click the **Add event** button in the **Scheduler panel**, or navigate to a day and time in the **Scheduler panel** and double-click when you want the programmed event to commence.



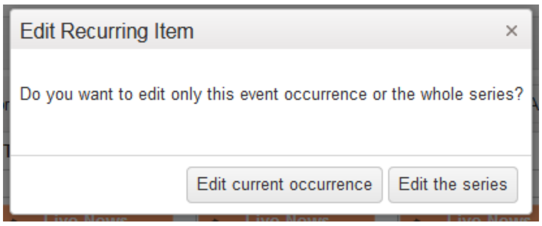
3. Add a **Title**, and adjust the **Start** and **End** time for each event. It is also possible to adjust the frequency of an event, e.g. daily, weekly or monthly.



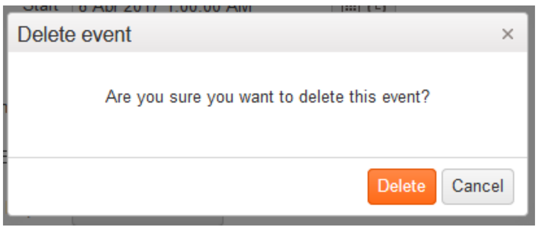
4. Click **Save** to add the new event to the **Scheduler panel**.

Edit or Delete an Event

1. Double-click an event displayed in the **Scheduler panel** to open the Event dialog.
2. Two options are presented if the event is a recurring event:
   - Edit the current event you have selected, or
   - Edit all events in the series.

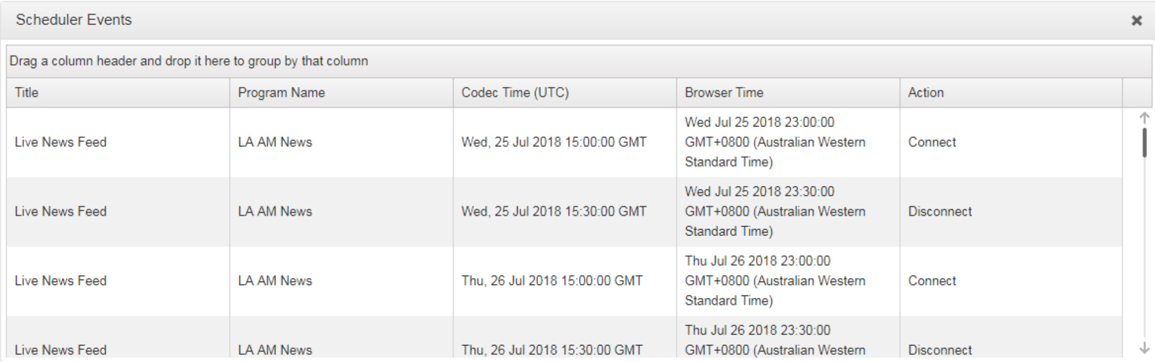


3. Select the preferred option and edit settings in the Event, then click **Save** to store all changes. Click **Delete** to delete an Event.
4. A confirmation dialog is presented to confirm the deletion of an event.



View Scheduler Events

To view all scheduled events click **Connect** in the **Menu Bar**, then select **Scheduler Events** to launch the **Scheduler Events panel**. Use the scroll bar to view future events.



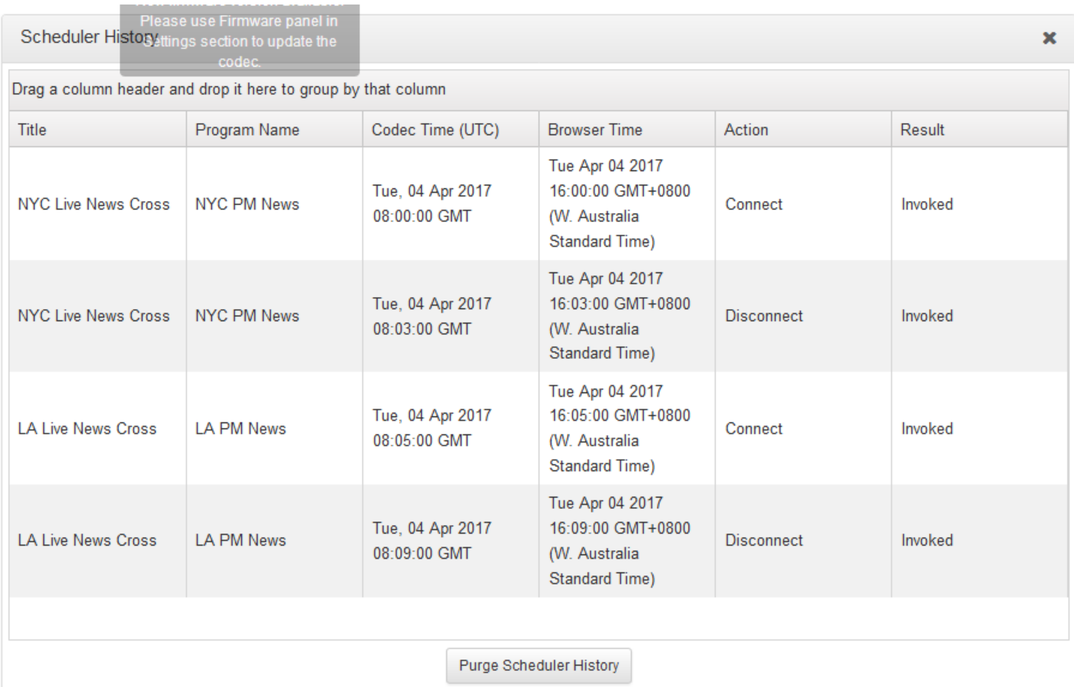
Scheduler Events

Drag a column header and drop it here to group by that column

Title	Program Name	Codec Time (UTC)	Browser Time	Action
Live News Feed	LA AM News	Wed, 25 Jul 2018 15:00:00 GMT	Wed Jul 25 2018 23:00:00 GMT+0800 (Australian Western Standard Time)	Connect
Live News Feed	LA AM News	Wed, 25 Jul 2018 15:30:00 GMT	Wed Jul 25 2018 23:30:00 GMT+0800 (Australian Western Standard Time)	Disconnect
Live News Feed	LA AM News	Thu, 26 Jul 2018 15:00:00 GMT	Thu Jul 26 2018 23:00:00 GMT+0800 (Australian Western Standard Time)	Connect
Live News Feed	LA AM News	Thu, 26 Jul 2018 15:30:00 GMT	Thu Jul 26 2018 23:30:00 GMT+0800 (Australian Western	Disconnect

View Event History

To view the history of scheduled events in a codec click **Connect** in the **Menu Bar**, then select **Scheduler History** to launch the **Scheduler History panel**. Press the **Purge Scheduler History** button to delete all events listed.



Scheduler History

Drag a column header and drop it here to group by that column

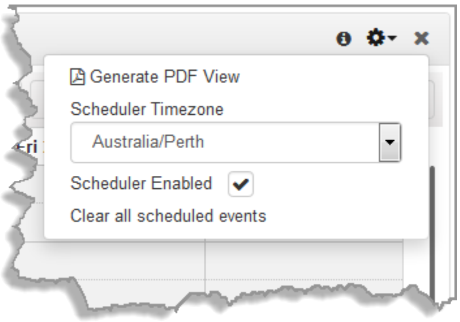
Title	Program Name	Codec Time (UTC)	Browser Time	Action	Result
NYC Live News Cross	NYC PM News	Tue, 04 Apr 2017 08:00:00 GMT	Tue Apr 04 2017 16:00:00 GMT+0800 (W. Australia Standard Time)	Connect	Invoked
NYC Live News Cross	NYC PM News	Tue, 04 Apr 2017 08:03:00 GMT	Tue Apr 04 2017 16:03:00 GMT+0800 (W. Australia Standard Time)	Disconnect	Invoked
LA Live News Cross	LA PM News	Tue, 04 Apr 2017 08:05:00 GMT	Tue Apr 04 2017 16:05:00 GMT+0800 (W. Australia Standard Time)	Connect	Invoked
LA Live News Cross	LA PM News	Tue, 04 Apr 2017 08:09:00 GMT	Tue Apr 04 2017 16:09:00 GMT+0800 (W. Australia Standard Time)	Disconnect	Invoked

Purge Scheduler History

Other Scheduler Options

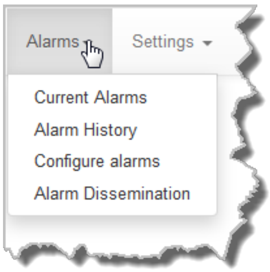
Click the **Options symbol** in the top right-hand corner of the panel to reveal a drop-down menu displaying other available options in the scheduler, including:

1. Generate a PDF view of scheduled events.
2. Adjust the Scheduler Timezone.
3. Enable / Disable the Scheduler.
4. Clear all Scheduled Events.



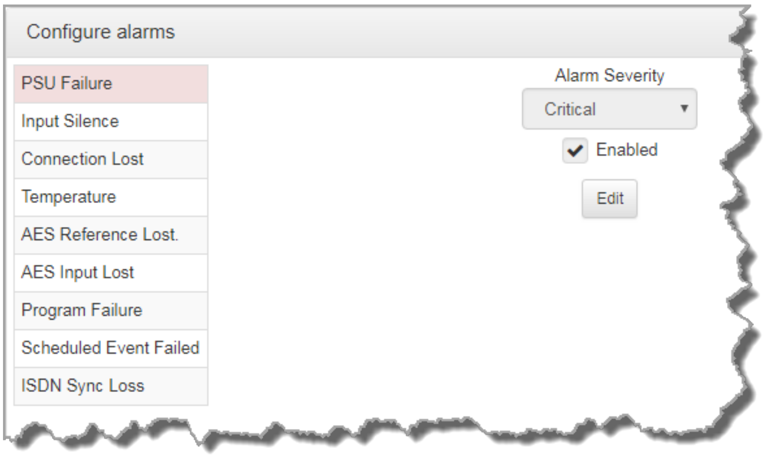
19.37 Configuring Alarms

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.

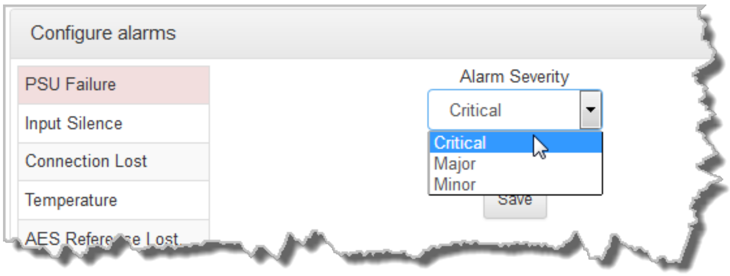


Configure and Enable Alarms

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then select **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.

Note: The following **System** and **Audio** alarms are available:

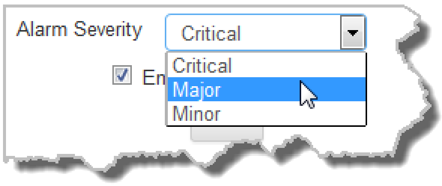
Alarm	Alarm Type	Explanation
PSU Failure	System	Raises an alarm if one or both PSUs fail
Input Silence	Audio	Raises an alarm if input audio is lost based on whether silence is detected on a single input, or pair of inputs (according to preconfigured silence detection threshold parameters)
Connection Lost	Audio	Triggers and alarm whenever a streaming connection is lost
Temperature	System	Raises an alarm if the temperature is too low or too high
AES Reference Lost	Audio	Raises an alarm if the AES reference clock signal is lost (not available in WheatNet-IP capable codecs)
AES Input Lost	Audio	Raises an alarm if the AES input signal is lost (not available in WheatNet-IP capable codecs)
Program Failure	Audio	An element within a configured program has failed and has not been terminated manually. Note: the alarm is cleared if the program is unloaded.
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.
ISDN Sync Loss	Audio	Raises an alarm if ISDN Sync is lost

Configuring Alarm Dissemination Severity Alerts

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.

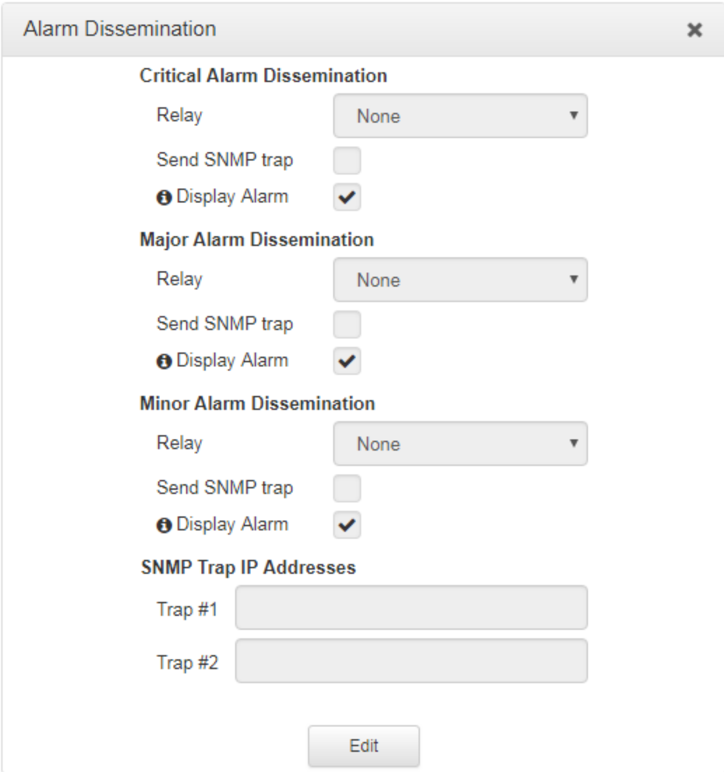


4. Click **Save** to store the new settings for the selected alarm.

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.



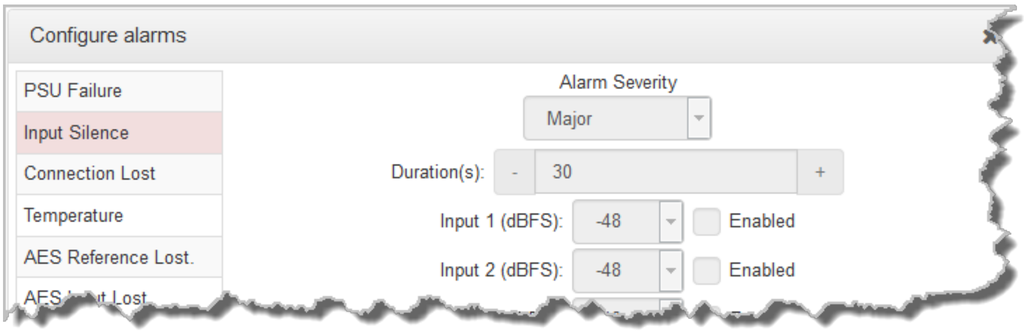
2. Click **Edit** to configure notification settings.
3. Select and configure relay, SNMP trap and alarm display settings for each **Alarm Severity** level. Enter **SNMP Trap IP Addresses** as required at the bottom of the panel.
4. Click **Save** to store the new settings.

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. See Configuring SNMP in the Codec for more info, or ask your system administrator.

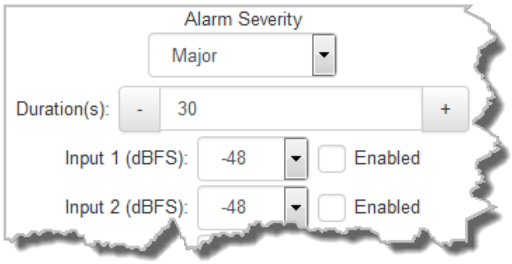
Configuring Input Silence Detection Parameters

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.



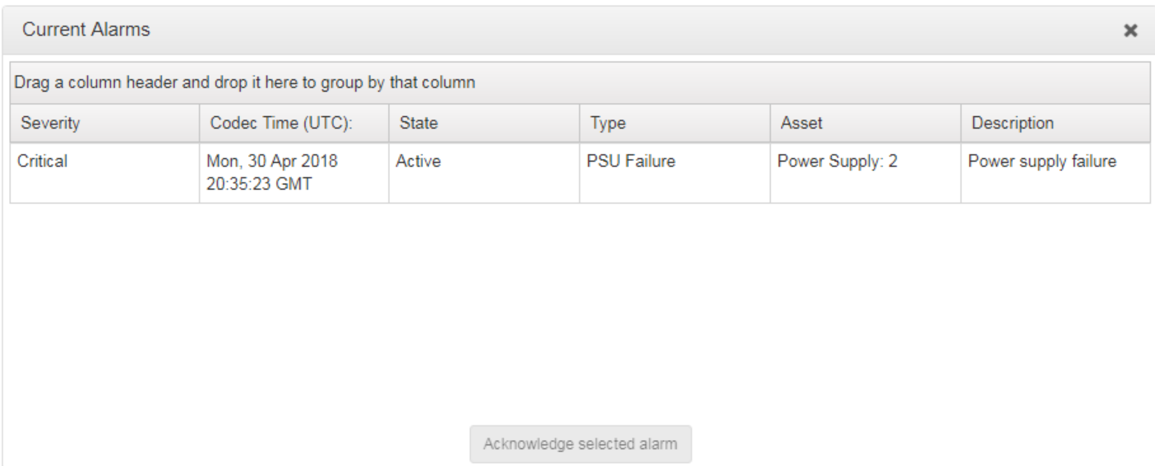
2. Click **Edit** to configure alarm settings.
3. Configure the dBFS threshold and timeout duration in seconds and ensure the input **Enabled** check-boxes are selected. An alarm will be raised when these thresholds are breached.



4. Click **Save** to store the new input silence alarm settings.

19.37.1 Managing Alarms

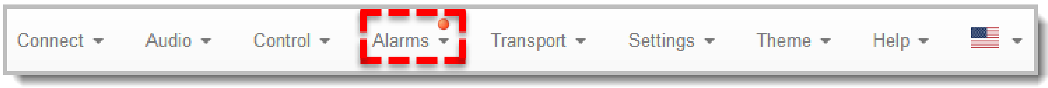
Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Current Alarms** to view active alarms.



Viewing Current Alarms

Active alarms are indicated by:

1. The red Alarm Symbol flashing in the toolbar of the HTML5 Toolbox Web-GUI screen.



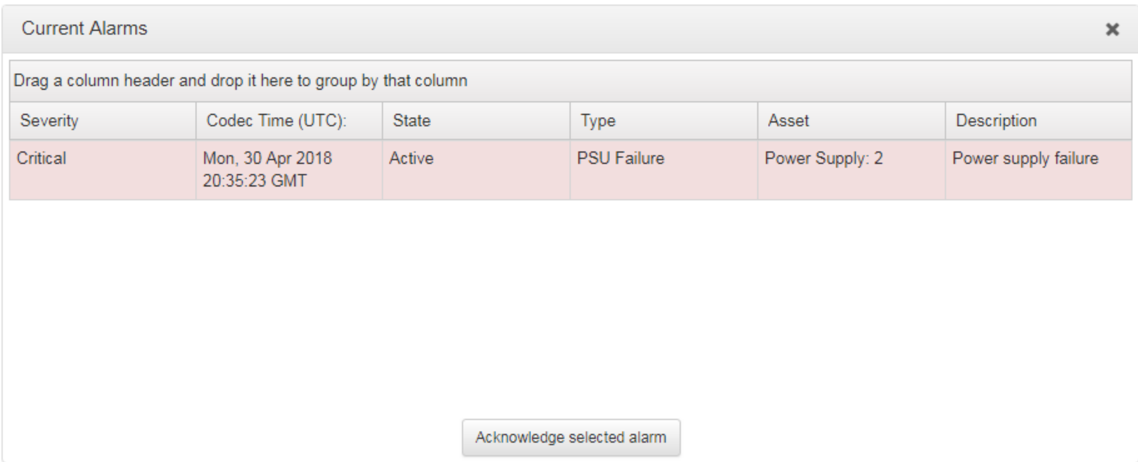
2. All new alarms being listed in the **Current Alarms panel**.
3. Other alerts as per **Alarm Dissemination panel** settings.
4. The codec front panel **ALARM LED** flashing red.

Important Note: When a connection is active the front panel **CONNECTED LED** is illuminated solid green. Illumination will 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 codec front panel **ALARM LED** will stop flashing and illuminate solid red.
4. The state of other alerts may change, as per **Alarm Dissemination panel** settings.

Alarm State	Front Panel Alarm LED	Web-GUI Alarm Symbol
Active	Flashing red	Flashing
Acknowledged	Solid red	Stops flashing, remains solid red

Deactivating Alarms

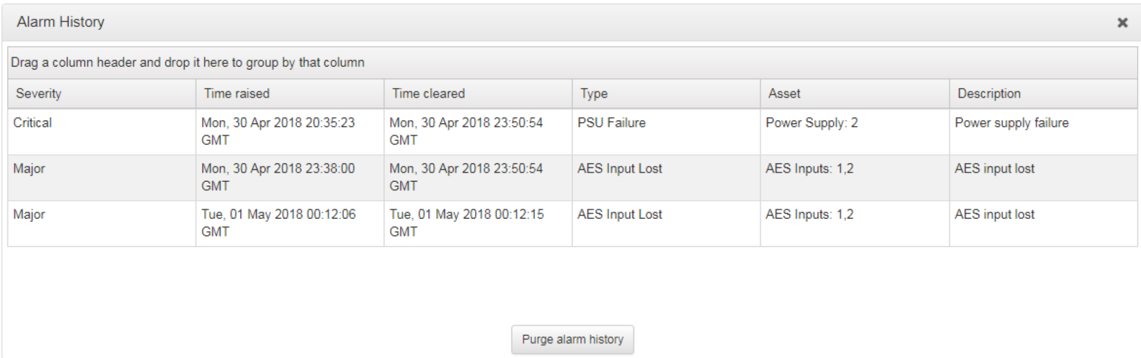
An alarm is deactivated automatically when the alarm state is reversed. E.g. if power is restored after a **PSU Failure** alarm, or if audio is restored after an **Input Silence** alarm.

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.



Severity	Time raised	Time cleared	Type	Asset	Description
Critical	Mon, 30 Apr 2018 20:35:23 GMT	Mon, 30 Apr 2018 23:50:54 GMT	PSU Failure	Power Supply: 2	Power supply failure
Major	Mon, 30 Apr 2018 23:38:00 GMT	Mon, 30 Apr 2018 23:50:54 GMT	AES Input Lost	AES Inputs: 1,2	AES input lost
Major	Tue, 01 May 2018 00:12:06 GMT	Tue, 01 May 2018 00:12:15 GMT	AES Input Lost	AES Inputs: 1,2	AES input lost

Click the **Purge alarm history** button to clear all alarms from the **Alarm History** panel.

19.38 RS232 Data Configuration

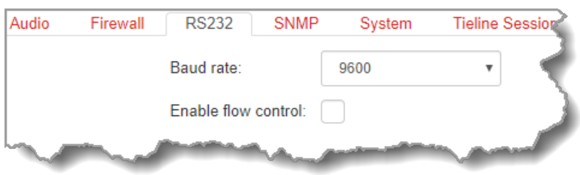
The codec supports both in-band and out-of-band data depending on the connection transport and algorithm you are using. In-band RPTP data is automatically included within an audio stream when using Tieline Music and Music PLUS algorithms over any transport – IP, ISDN or POTS. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP	ISDN and POTS
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 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	• In-band RPTP data is enabled automatically and used for all rules including relay reflection
All other algorithms	• No in-band data available; synchronized out-of-band data can be enabled and disabled	• No in-band or out-of-band data available

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 (see Enabling RS232 Data).

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 to G3 codecs over IP, ISDN or POTS only in-band data is available via the Music and MusicPLUS algorithms.
- Use firmware higher than 2.8.xx in the 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.

19.39 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 WheatNet-IP logic IO, or a codec program being connected or disconnected. There are three categories of rules:

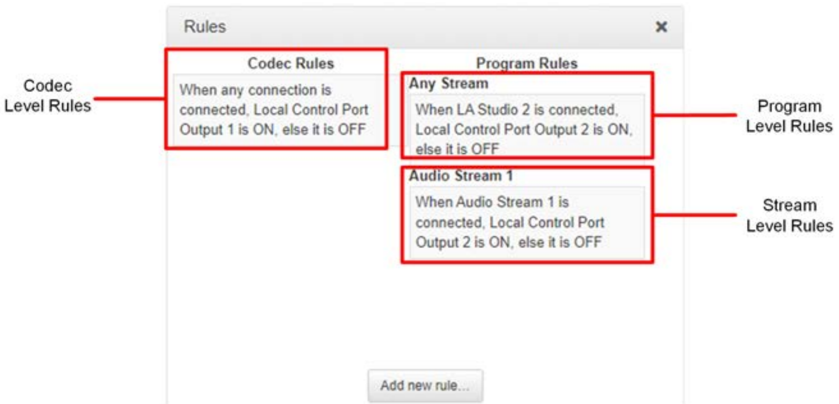
1. Codec level rules: Rules based on programs or codec hardware and software I/O states, e.g. Connect or disconnect a program when an input is toggled, or synchronize a local input to a remote relay.
2. Program level rules: Rules based on codec behaviors at the program level, e.g. Connect and disconnect program A when an input is toggled, set a custom mix when a relay is activated, or synchronize a local input to a remote relay.
3. Stream level rules: Rules based on codec behaviors at the stream level, e.g. Connect and disconnect stream A when an input is toggled, or 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:

1. Press the **HOME** button to return to the **Home** screen
2. Use the navigation buttons on the front panel to select **Connect** and press the button
3. Select **IP** and press the button
4. Select your preferred **IP Mode** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Data** and press to toggle between **Enabled** and **Disabled**.

For more information please see Enabling Relays & RS232 Data.

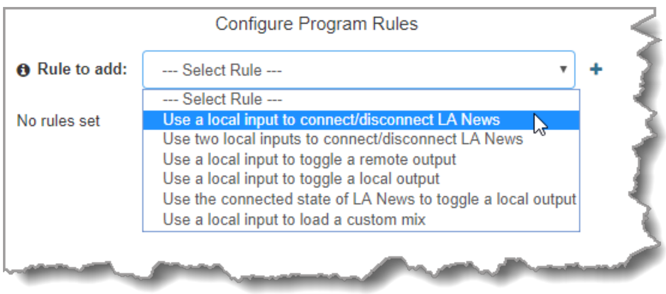
Important Notes for Rules:
- The codec has 4 physical **CONTROL PORT GPIOs**; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

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

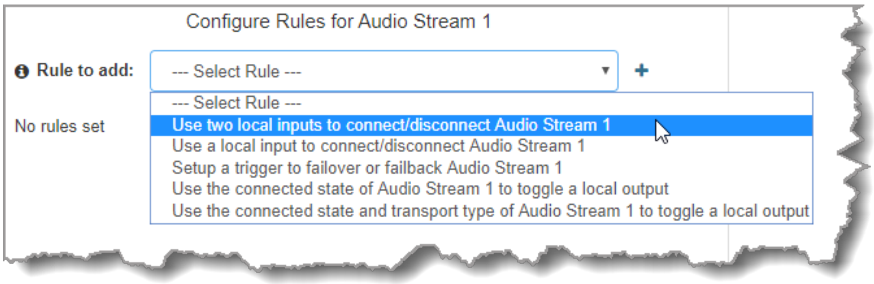
In the **Program Manager panel** wizard use the **Configure Program Rules** screen to configure program level rules. The rules available are displayed in the following image.



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.

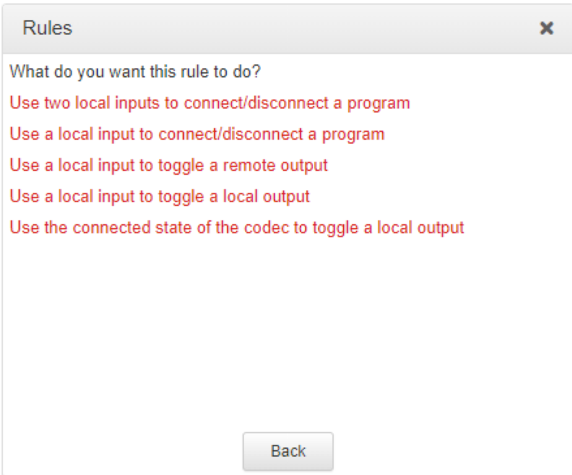


Note: A subset of filtered rules will be displayed for an **Answer only** audio stream connections.

Configuring Rules with 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.

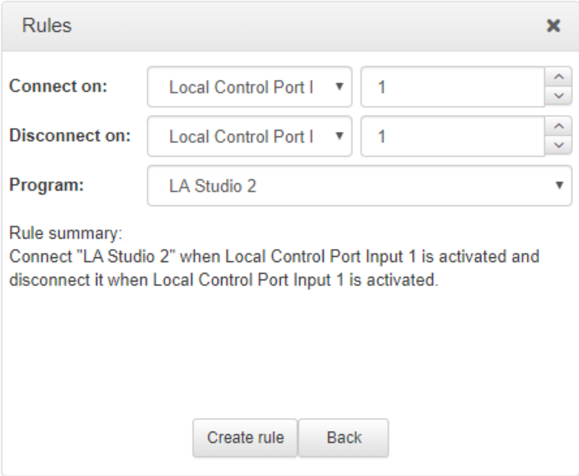


Note: When rules have been configured previously they are displayed when the **Rules panel** is opened.

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 first 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.

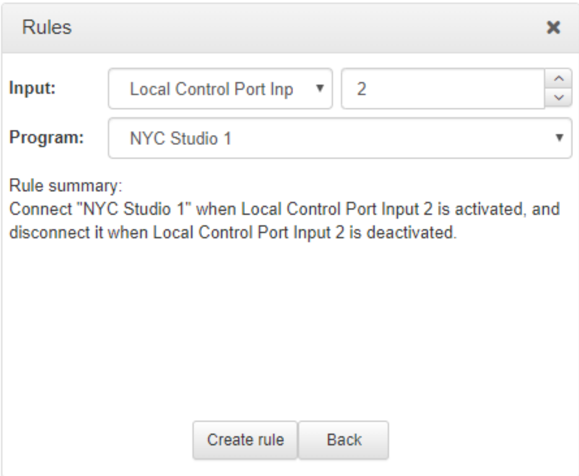


4. Check the **Rule summary** and click **Create Rule** to save the settings.

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 second 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.

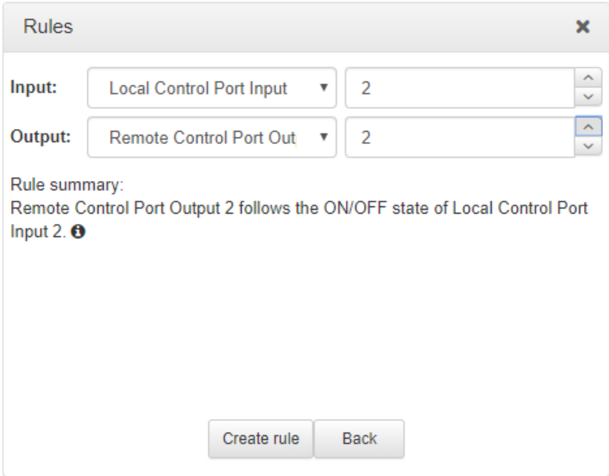


4. Check the **Rule summary** and click **Create Rule** to save the settings.

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.

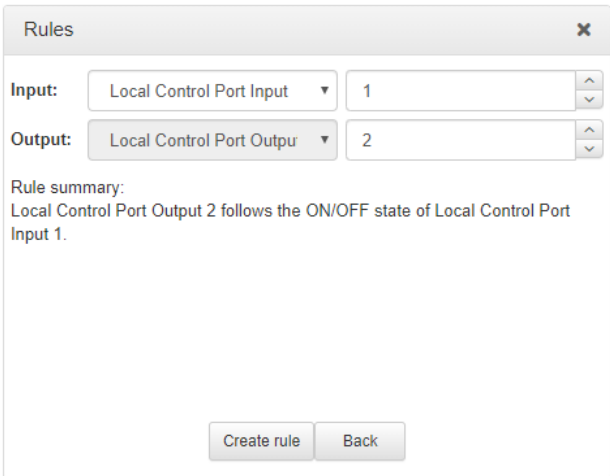


3. Check the **Rule summary** and click **Create Rule** to save the settings.

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.

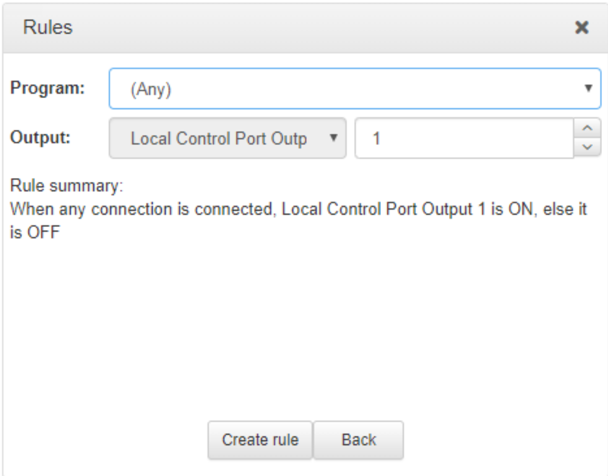


3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 5: Use the Connected State of the Codec to Toggle a Local Output

This rule is used to toggle a codec's control port relay output each time a program connects and disconnects.

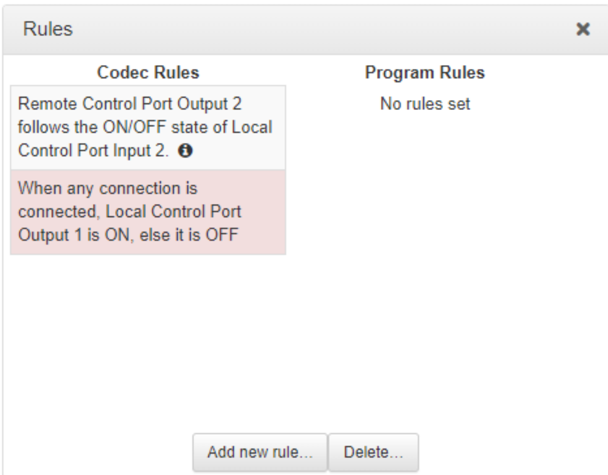
1. Click the fourth rule in the **Rules panel** titled **Use the connected state of the codec to toggle a local output**.
2. Click the drop-down **Program** arrow to select the program which will affect the relay toggle function, or use the default setting whereby any program connecting will toggle the 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.

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.

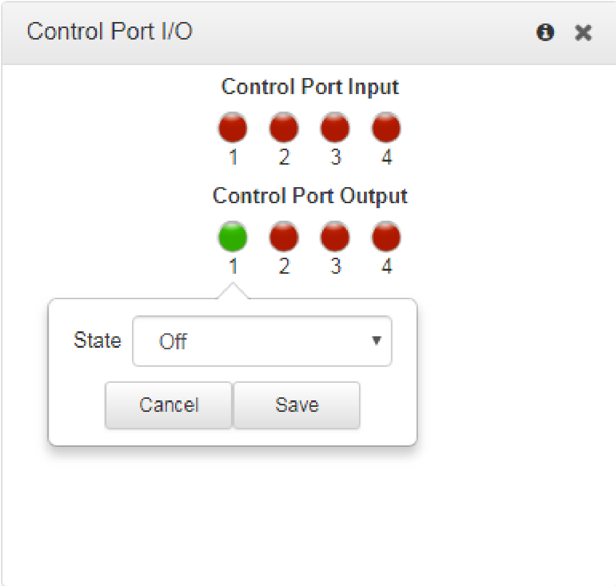


4. Click **Yes** in the confirmation dialog.

19.40 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 using the DB15 **CONTROL PORT IN/OUT** 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.



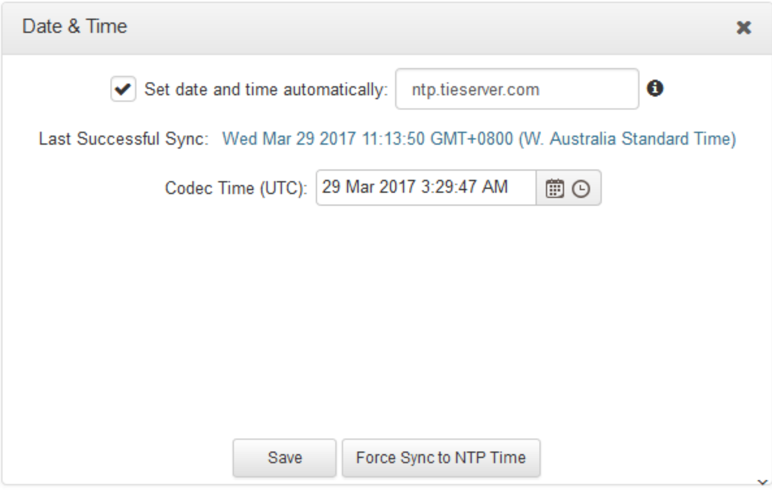
Important Note: Tieline and WheatNet virtual logic inputs 5-7 can be activated by pressing the **F1** button and keypad numbers 1-3 on Genie and Merlin codecs.

19.41 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.

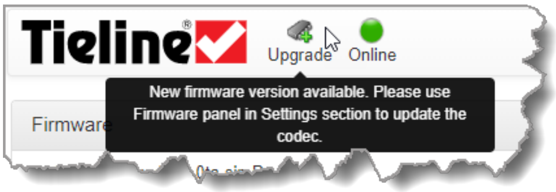
19.42 Upgrading Codec Firmware

To download the latest codec firmware visit www.tieline.com. See Upgrading Firmware via USB to upgrade codec firmware using a USB stick 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 a LAN 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.

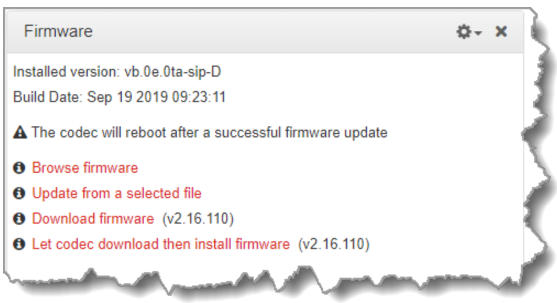
Performing a Firmware Upgrade

There are several firmware upgrade options available:

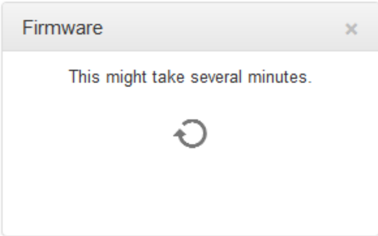
1. **Browse Firmware**: Click to navigate to the Tieline website and download the latest firmware for the codec.
2. **Update from a Selected file**: Click to navigate to a firmware file saved on a computer or network drive.
3. **Download firmware**: Click to download a previous reliable firmware version. 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**.



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 you are using to perform 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.



5. We recommend 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

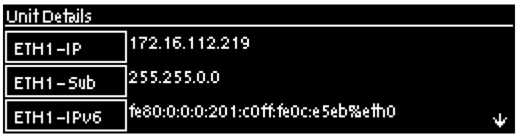
20 Front Panel Configuration Tasks

The following sections explain how to configure codec settings using the front panel **LCD SCREEN** and **KEYPAD**.

20.1 Configuring IP via the Front Panel

Checking IP Address Details in the Codec

1. Press the **SETTINGS** button.
2. Select **Unit** and press the button.
3. IP address details and other unit details are listed. Use the arrow up and down 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, Wi-Fi and VLAN Configuration Options

The codec features two physical Ethernet port interfaces and up to four additional VLAN interfaces. The codec also supports a Wi-Fi connection via the USB port on the front panel.

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

As an example, if only one physical Ethernet interface is available, Wi-Fi or VLANs can be used to operate SmartStream PLUS, or to separate codec Control and Streaming functions if required.

Following are a range of options which can be configured in the **Network** menu. After completing configuration ensure you navigate to **Apply Setting** and press the button to apply the new settings.

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).

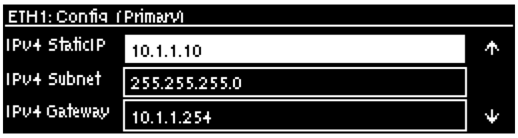
1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **ETH1**, **ETH2**, **Wi-Fi** or a **VLAN** interface.

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 Mode** 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.

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **ETH1**, **ETH2**, **Wi-Fi** 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 Mode** and press the button.
6. Select **Static** and press the button.
7. Navigate to **IPv4 Static** and enter the IP address, then press the button.
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the button.
9. Navigate to **IPv4 Gateway** 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 each Ethernet port, a Wi-Fi connection via the **USB PORT** on the codec and any VLANs which are 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 **ETH1**, **ETH2**, **Wi-Fi** or a **VLAN** interface.
4. Select **Config** and then **IPv6 Mode** 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 **IPv6 Static** (Address)**, IPv6 Prefix** and IPv6 **Gateway** fields in the codec to manually configure address details.

DNS Server

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.

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **ETH1**, **ETH2**, **Wi-Fi** or a **VLAN** interface, then 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 interface will operate 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 **ETH1**, **ETH2** 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 and press the button.
4. Navigate to **Config** and press the button.
5. Select **Usage** and press the button.
6. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
7. Use the down navigation button to scroll to **VLAN ID**.
8. Press the button to enter a number between 1-4094 inclusive.
9. Press the button to confirm this setting.
10. 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 and press the button.
4. Navigate to **Config** and press the button.
5. Select **Usage** and press the button.
6. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
7. Use the down navigation button to scroll to **VLAN Priority**.
8. Press the button to enter a number from 0 to 7 inclusive.
9. Press the button to confirm this setting.
10. 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)

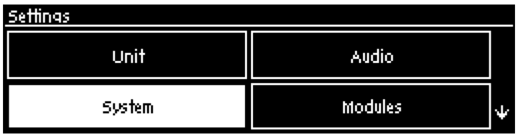
This setting applies the VLAN settings to a physical Ethernet port in the codec.

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 and press the button.
4. Navigate to **Config** and press the button
5. Select **Usage** and press the button.
6. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
7. Use the down navigation button to scroll to **Interface**.
8. Press the button to select **ETH1** or **ETH2**, then press the button.
9. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

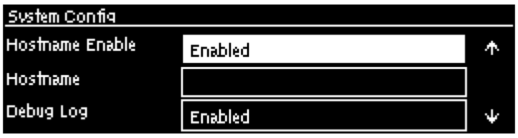
20.2 Configuring a Hostname

It is possible to assign a hostname to the codec to provide a flexible way of identifying the codec on a network.

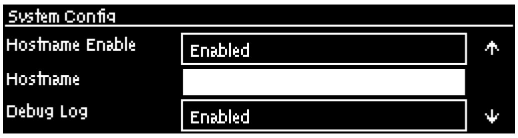
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.

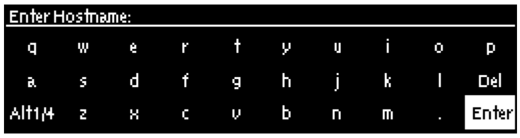


3. Navigate down to **Hostname Enable** and press the button to enable this feature.



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.





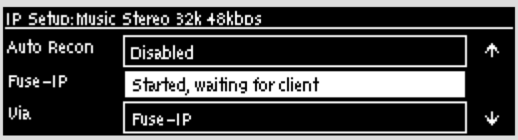
Important Note:
- Modifying hostname settings requires a codec restart before they take effect
- In the **Hostname** only enter the characters a-z, A-Z, 0-9 and - and the first or last character cannot be a hyphen/dash.

20.3 IP Via Setting

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 **Via** interfaces in order of use when available are:

1. **ETH1** Ethernet port (default **Primary** Via interface)
2. **ETH2** Ethernet port (default **Secondary** Via interface)
3. **Wi-Fi** (default **Tertiary** Via interface)

Important Notes:
- If an interface is not available it is not listed in the **Via** interface selection screen. E.g. if Wi-Fi is not configured it will be unavailable.
- If you select Fuse-IP as the **Via** interface, a link to the Fuse-IP configuration menu appears above the **Via** menu.

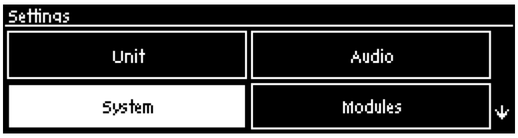


- VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.

Reconfigure Default Primary, Secondary and Tertiary Interfaces

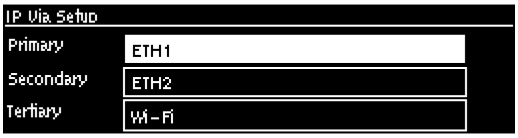
It is possible to reconfigure the default **Primary** (Ethernet 1), **Secondary** (Ethernet 2) and **Tertiary** (Wi-Fi) interfaces in the codec. As an example, you may want to select **Primary** as the dialing interface in a program and then copy this program onto multiple codecs. However, the actual 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.

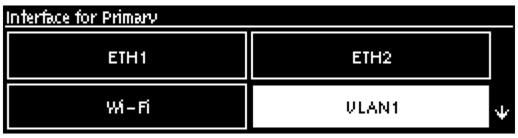


3. Navigate to **IP Via Setup** and press the button.

4. Navigate to the **Primary**, **Secondary** or **Tertiary** IP Via interface setting and press the button.



5. Select an alternative default interface and press the button.



6. Press the **Return button** to navigate back 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 or Tertiary **Via**.

20.4 Connecting over Wi-Fi

The codec supports connecting over Wi-Fi using a cellular USB dongle attached to the USB port on the front panel. A Wi-Fi connection can be used for codec Control and/or Streaming.

Important Notes:

- Codecs using Wi-Fi must have firmware version 2.16.xx or higher installed.
- The list of cellular Wi-Fi dongles that have been used successfully are available on the Tieline website at www.tieline.com.
- If you are leaving codecs unattended with a Wi-Fi dongle attached, reboot the codec with the dongle in it to confirm there is no issue rebooting after an intermittent loss of power.

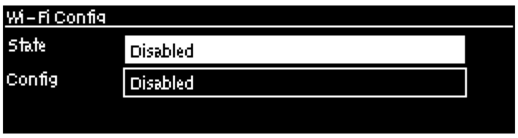
Enabling Wi-Fi

To enable Wi-Fi:

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

3. Navigate to **Wi-Fi** and attach a compatible cellular Wi-Fi dongle to the USB port on the codec. By default the Wi-Fi display should change from **Not Present** to **Disabled**. Press the button to enable Wi-Fi in the **State** menu. Note: If you attach a USB dongle when Wi-Fi has already been enabled, or you attach a new USB dongle, you may need to disable Wi-Fi and then enable it for the codec to recognize the USB when it is first attached.

4. Navigate to **State** and press the button to select **Yes** and enable **Wi-Fi**.



5. The codec screen should display an "Enabling Wi-Fi" and "Acquiring Address" status message before Wi-Fi is enabled.

Connecting to a Wi-Fi Network

1. Navigate to **Rescan** in the **Wi-Fi Config** menu and press the button to refresh the Wi-Fi networks visible. Note: Signal strength is indicated by a fractional number out of 4. A signal strength of 2/4 or higher is recommended for reliable streaming.

2. Navigate down to the preferred Wi-Fi network and press the button to select it.

3. Navigate to **Password**, then press the button. Use the codec's navigation buttons to select the alphanumeric characters on the **LCD SCREEN** and enter the network password. Press the **Enter** button on the **LCD SCREEN** to complete password configuration.

4. The password is now entered. Navigate up to **Save & Connect to Wi-Fi Network** and press the button to connect to the network.

5. Once connected the codec can dial and connect using IP over Wi-Fi.

Adding a Wi-Fi Custom Access Point

Some Wi-Fi Networks have hidden SSIDs and will not automatically appear in the list of available networks after performing a network scan. To connect to these networks they need to be manually added to the codec:

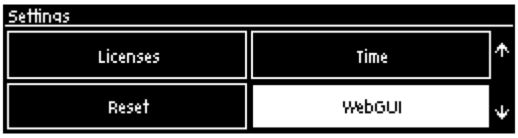
1. In the **Wi-Fi Config** menu navigate down to **Add Wi-Fi Network** and press the button.

2. Select and enter the details for the network **SSID** and **Password**, then navigate up to **Save and Connect to Wi-Fi Network** and press the button to join the hidden Wi-Fi network.

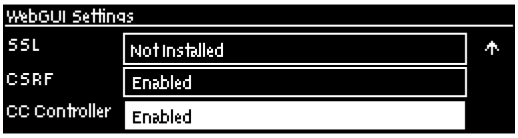
20.5 Enabling the Cloud Codec Controller

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**.



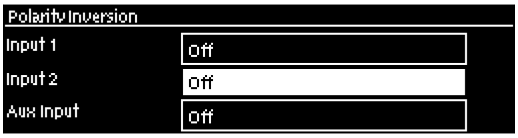
Important Notes:
- Ensure **CSRF** is disabled in the codec or it will not be able to connect to the CCC. This setting is **[OFF]** by default and is also available in the codec menu via **Settings > WebGUI**, and in the **Options** panel in Toolbox.
- Locally Defined Codecs over a private network do 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.

20.6 Inverting Input Polarity

The codec supports inverting input polarity.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Audio** and press the button.
3. Navigate down to **Polarity Inv** and press the button.

4. Select the input for which polarity is to be inverted and press the button to toggle between **Off** and **On**.



20.7 Connecting with 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, which may include:

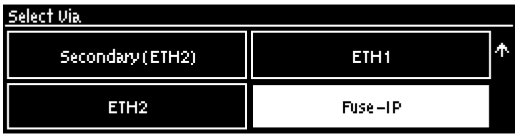
- Dual Ethernet LAN ports.
- An Ethernet LAN port with a Wi-Fi dongle.

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 **Via** interface you can use to dial, similar to selecting a LAN port or Wi-Fi.



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 easier to dial static IP addresses configured at the studio than cellular, Wi-Fi or LAN interfaces at the remote site. Like SmartStream PLUS redundant streaming, you can use two IP interfaces at the studio for additional redundancy.

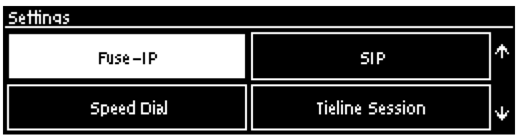
Prerequisites

Version 2.16.xx firmware is required to use Fuse-IP. Before configuring Fuse-IP you need:

- 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.

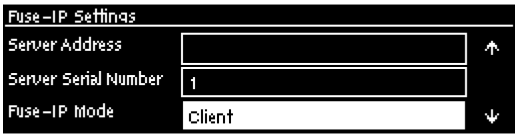
Configuring a Fuse-IP Server at the Studio

1. Press the **SETTINGS** button.
2. Select **Fuse-IP** and press the button.

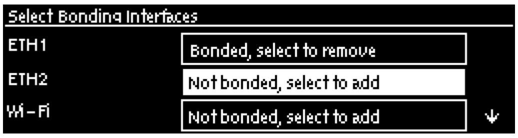


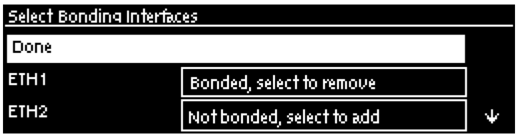
Important Note: Ensure Fuse-IP is disabled prior to configuration. Navigate to **Fuse-IP Status** and press the button to disable Fuse-IP if it is enabled.

3. Navigate down to **Fuse-IP Mode** and press the button to select **Server** if the codec is at the studio. Note: the server codec serial number will be displayed and needs to be entered into the Fuse-IP client codec.



4. Navigate to **Bonded Interfaces** and press the button, then navigate to each interface in turn and press the button to select or deselect interfaces. Navigate to the top of the menu after configuration and select **Done**, then press the button.





5. 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.

6. The codec is now configured to connect with a client codec over a **Fuse-IP** tunnel.

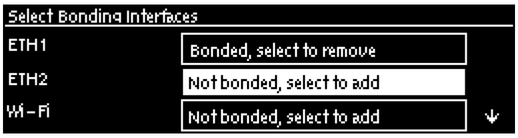
Configuring a Fuse-IP Remote Client

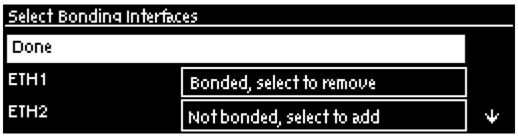
1. Press the **SETTINGS** button.
2. Select **Fuse-IP** and press the button.

Important Note: Ensure Fuse-IP is disabled prior to configuration. Navigate to **Fuse-IP Status** and press the button to disable Fuse-IP if it is enabled.

3. Navigate to **Fuse-IP Mode** and press the button to select **Client** if it is the remote codec.

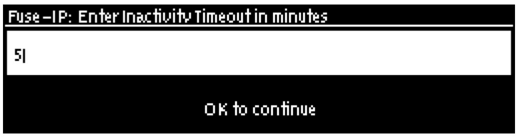
4. Navigate to **Bonded Interfaces** and press the button, then navigate to each interface in turn and press the button to select or deselect interfaces. Navigate to the bottom of the menu after configuration and select **Done**, then press the button.





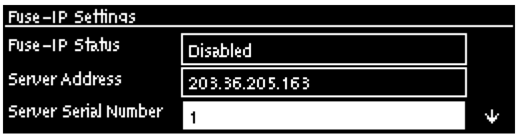
5. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be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

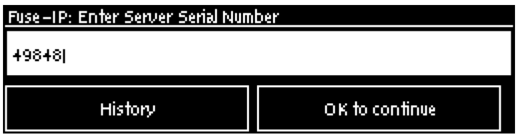
6. Navigate to **Inactivity Timeout** and press the button if you want to adjust the predetermined time period for turning the Fuse-IP tunnel off. Adjust the setting and press the button to store the new setting. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter 0 to disable the timeout.



7. Navigate to **Server Address** and press the button to enter a public static IP address associated with the bonded interfaces at the studio (or select a previously entered server address via **History**), then press the 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.

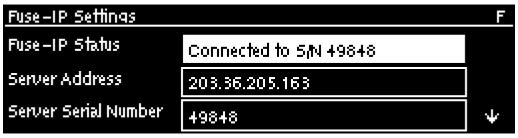
8. Navigate to **Server Serial Number** and press the button to enter the serial number of the server codec to which you are connecting (or select a previously entered serial number via History), then press the button.





9. Navigate up to Fuse-IP Status and then press the button to create a Fuse-IP tunnel between the server and client codecs.

10. The letter **F** is displayed on the far right side of the **Status Bar** to confirm the two codecs have created a Fuse-IP tunnel. Remember Fuse-IP must be enabled on both codecs.



Please note: double-check all settings on both the server and client codecs if the message **Started, dialing server** persists after starting Fuse-IP.

11. Select **Fuse-IP** as the **Via** 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.

20.8 Selecting an Algorithm

The codec offers PCM uncompressed linear audio as well as 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. There is a range of pre-programmed connection profiles to simplify codec configuration. See Choosing Dialing Profiles for more details.

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 optimized 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 optimize audio quality. If lower bandwidth 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 optimized 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.

aptX® Enhanced is supported over ISDN at the following sample and bit rates:

Encoding	Bit rate Required	B Channels Required
aptX® Enhanced Mono 16 bit, 32 kHz	128 kbps	2
aptX® Enhanced Mono 16 bit, 48 kHz	192 kbps	3
aptX® Enhanced Mono 24 bit, 32 kHz	192 kbps	3
aptX® Enhanced Stereo 16 bit, 32 kHz	256 kbps	4

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 algorithm 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 **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Algorithm** and press .
7. Navigate to **Manual** to configure all settings manually, or **Profile** to select a pre-configured algorithm profile, then press .

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 remotes into a broadcast. The algorithm you select to connect with will also depend upon:

- The codecs to which you are connecting (Tieline versus non-Tieline)
- Whether you are creating multi-unicast connections.
- Whether you are connecting using SIP or not.
- 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 which is the best algorithm setting for the connection you are setting up. Please see the following table for details on the connection requirements of the different algorithms available.

Algor-ithm	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
Linear/PCM (Uncom-pressed)	16/24 bit up to 45kHz	0ms	sample rate x bits per sample x no. channels; 512kbps minimum (16bit;32kHz) to 4.6 Mbps (24bit; 96 kHz)	80kbps	• Full bandwidth, perfect audio quality for voice and music • No error concealment/correction or artifacts	• 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. POTS or 3G wireless) • Suitable when bidirectional communication between announcers is required • Deliver 15kHz stereo over 1 x 64kbps ISDN B Channel.
Tieline Music-PLUS	Up to 22kHz	20ms	48 kbps minimum (Optimized 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
AAC-ELDv.2	Up to 20kHz	35ms	Pending release	Pending release	• High quality voice and music • Low delay at low bit rates	• Great for live voice or music remotes where limited connection bandwidth is available • Suitable when bidirectional communication between announcers is required
aptX Enhanced	10Hz-24kHz	2.5ms at 48kHz	128kbps minimum (16bit; 32kHz)	80kbps	• Very high quality voice and music	• Ideal for STLs and audio distribution where high

			to 288kbps (24bit;48kHz)		• Extremely low delay at high bit rates • Highly cascade resilient	connection bandw idth is available and very low delay is highly desirable. • Resilient w ith multiple encodes/decodes w hen 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
TxTran / RxTran					NOT FOR BROADCAST USE	NOT FOR BROADCAST USE

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.

IP Connection Bit rates Supported

Algorithm	Sample Rate	Genie and Merlin Connection Bit Rates Supported over IP																										
AAC HE Mono	32kHz				16	20		24		32	40	48	56															
AAC HE Mono	44.1kHz							24		32	40	48	56	64														
AAC HE Mono	48kHz							24		32	40	48	56	64														
AAC HE Stereo	32kHz							24		32	40	48																
AAC HE Stereo	44.1kHz										40	48	56	64														
AAC HE Stereo	48kHz										40	48	56	64														
AAC LD Mono	32kHz										40	48	56	64														
AAC LD Mono	44.1kHz												56	64	96	112	128	160		192								
AAC LD Mono	48kHz												56	64	96	112	128	160		192								
AAC LD Stereo	32kHz													64	96	112	128											
AAC LD Stereo	44.1kHz														96	112	128	160		192	256			320		384		
AAC LD Stereo	48kHz														96	112	128	160		192	256			320		384		
AAC ELD Mono	32kHz							24		32	40	48	56	64														
AAC ELD Mono	44.1kHz							24		32	40	48	56	64														
AAC ELD Mono	48kHz							24		32	40	48	56	64														
AAC ELD Stereo	32kHz											48		64	96	112	128											
AAC ELD Stereo	44.1kHz											48	56	64	96	112	128											
AAC ELD Stereo	48kHz											48	56	64	96	112	128											
AAC LC Mono	32kHz									32	40	48																
AAC LC Mono	44.1kHz											48	56	64	96		128	160										
AAC LC Mono	48kHz											48	56	64	96		128	160										
AAC LC Stereo	32kHz													64	96													
AAC LC Stereo	44.1kHz													64	96	112	128	160		192	256			320				
AAC LC Stereo	48kHz													64	96	112	128	160		192	256			320				
MP2 Mono	24kHz													64														
MP2 Mono	32kHz													64	96	112	128	160		192								
MP2 Mono	44.1kHz													64	96	112	128	160		192								
MP2 Mono	48kHz													64	96	112	128	160		192								
MP2 Stereo	32kHz															112	128	160		192	256							
MP2 Stereo	44.1kHz															112	128	160		192	256			320		384		
MP2 Stereo	48kHz															112	128	160		192	256			320		384		
MP2 J-Stereo	32kHz															112	128			192								
MP2 J-Stereo	44.1kHz															112	128			192								
MP2 J-Stereo	48kHz															112	128			192								
MP3 Mono	32kHz													64	96	112	128	160										
MP3 Mono	44.1kHz													64	96	112	128	160										
MP3 Mono	48kHz													64	96	112	128	160										
MP3 Stereo	32kHz															112	128	160		192	256							
MP3 Stereo	44.1kHz																128	160		192	256			320				
MP3 Stereo	48kHz																128	160		192	256			320				
aptX Enhanced Mono	32kHz																128			192								
aptX Enhanced Mono	44.1kHz																		176.4			264.6						
aptX Enhanced Mono	48kHz																			192			288					
aptX Enhanced Stereo	32kHz																				256					384		
aptX Enhanced Stereo	44.1kHz																								352.8		529.2	
aptX Enhanced Stereo	48kHz																									384		576
Opus Voice	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128											
Opus Mono	48kHz											48		64	96	112	128											
Opus Stereo	48kHz													64	96	112	128	160		192	256							
Music Mono	32kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music Stereo	32kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music PLUS Mono	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music PLUS Stereo	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							

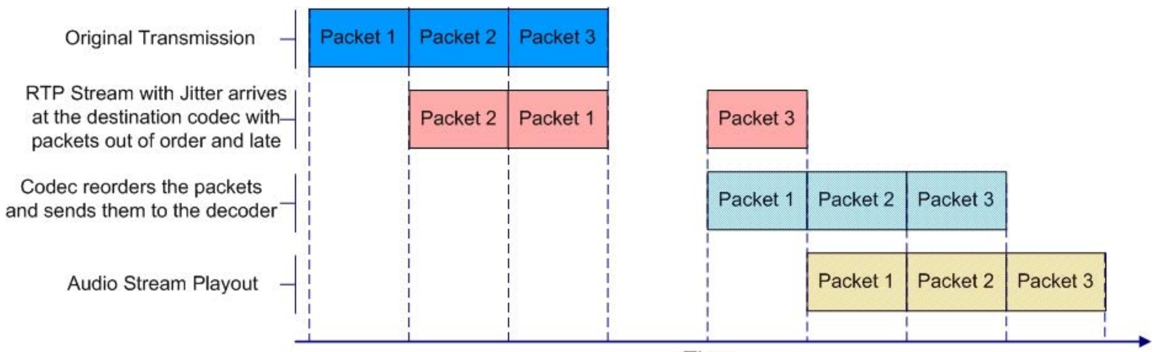
ISDN Encoding Options

The codec supports ISDN connections using the following algorithms and B Channel assignments.

ISDN Encoding	1B	2B	3B	4B
E-AptX Mono 16bit 32KHz		✓		
E-AptX Mono 16bit 48KHz			✓	
E-AptX Mono 24bit 32KHz			✓	
E-AptX Stereo 16bit 32KHz				✓
Music Mono	✓	✓	✓	✓
Music Stereo	✓	✓	✓	✓
Music Plus Mono	✓	✓	✓	✓
Music Plus Stereo	✓	✓	✓	✓
MP2 Mono 32KHz	✓	✓		
MP2 Stereo 32KHz		✓		
MP2 Mono 48KHz	✓	✓		
MP2 Stereo 48KHz		✓		
MP2 J-Stereo 32KHz		✓		
MP2 J-Stereo 48KHz		✓		
G.711	✓			
G.722	✓			

20.9 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).

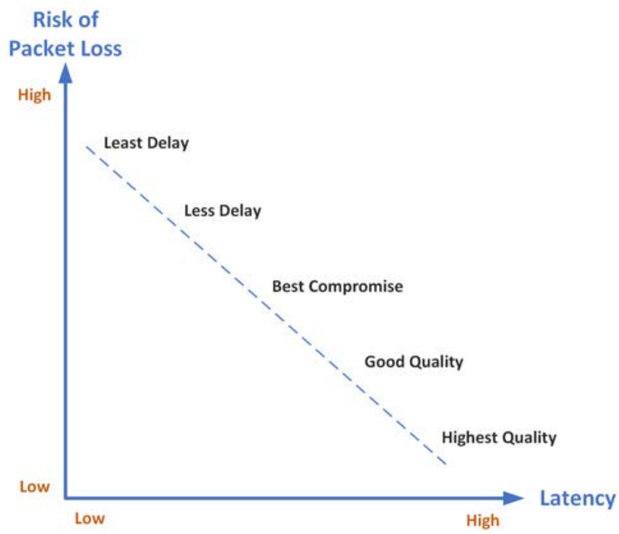
With Tieline codecs you can configure either a fixed or automatic jitter buffer and the settings you use depend on the IP network over which you are connecting. Over LANs, WANs and wireless networks the automatic jitter buffer generally works well. It adapts automatically to prevailing IP network conditions to provide continuity of audio streaming and minimize 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 programmed 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 Support 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
Linear (Uncompressed)	✗	✗
Tieline Music	✓	✗
Tieline MusicPLUS	✓	✗
G.711	✗	✓
G.722	✗	✓
MPEG Layer 2	✓	✓
MPEG Layer 3	✓	✗
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 **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.

6. Navigate to **Jitter Buffer** and press .
7. Select **Auto Jitter Adapt** and press .
8. Select your preferred jitter buffer setting and press .

Important Notes: There is no jitter buffer setting on a multicast server codec because it only sends and never receives audio packets.

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 allow the codec to evaluate 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 adjusted to suit prevailing 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 jitter buffer states. Jitter buffer and connection status statistics can be viewed via **HOME** **> Cxns** and use the down and up navigation buttons to scroll through connection statistics. The first four stages are observed in “auto” jitter buffer mode.

1. **Stabilization period (a1):** A few seconds during which a stable connection is established.
2. **Stage 2 (a2):** A compatibility check occurs.
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.

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 **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Jitter Buffer** and press .
7. Select **Fixed Buffer Level** and press .
8. Use the numeric **KEYPAD** to enter the fixed buffer value in milliseconds and press . 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 programming 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 (linear 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 lost data packets. 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 FEC is employed. We recommend you add 100ms to the fixed 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 when this feature is enabled.

20.10 Configuring Forward Error Correction

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

1. Tieline FEC: Transmits a secondary stream of audio data packets over a single connection.
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	Tieline 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 to the connection bit rate setting in the codec.	Recommended for wired LAN connections & managed T1 & E1 connections for STLs with connections that aren't shared & have quality of service (QoS).	Yes	Yes

Configuring Tieline FEC 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 **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **FEC** and press .
7. Select the local codec FEC setting in the **Local FEC** screen and press .
8. Select the remote codec FEC setting in the **Remote FEC** screen and press .

Important Note: FEC can only be configured for use with the Music and MusicPLUS algorithms.

Configuring RFC 2733 FEC in the Codec (Sessionless Connections only)

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Sessionless** and press the button.
5. Select **Peer-to-Peer** and press the button.
6. Use the down navigation button to select **Setup** and press the button.
7. Navigate to **FEC** and press .
8. Select **Enable** and press .
9. Select the **FEC Percentage** and press .
10. Select the **FEC Delay** used by the codec and press .

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.

20.11 Configuring Encode/Decode Direction

By default the codec by is configured to both encode and decode data. However, it is possible to configure the codec to either encode or decode audio data only. This is useful for:

- Conserving connection bandwidth when unidirectional data streaming is required.
- Lowering data costs.
- Increasing overall connection reliability.

Configure the transmitting codec to encode only and program the receive codec to decode only when using this feature. 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 **Connect** and press the button.
3. Select **IP** and press .
4. Select your preferred **IP Session** mode and press the button.
5. Use the down navigation button to select **Setup** and press .
6. Navigate to **Dir** and press .
7. Select **Encode Only** or **Decode Only** and press .

20.12 Enabling Relays & RS232 Data

Data must be enabled to activate contact **CONTROL PORT** 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 any transport – IP, ISDN or POTS. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP	ISDN and POTS
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 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	• In-band RPTP data is enabled automatically and used for all rules including relay reflection
All other algorithms	• No in-band data available; synchronized out-of-band data can be enabled and disabled	• No in-band or out-of-band data available

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 on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select **Tieline** (or **Sessionless** and then **Peer-to-Peer**) and press the button.
5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to **Data** and press to toggle between **Enabled** and **Disabled** (Note: default setting is **Disabled**)

Please see Appendix A for RS232 and Control Port Wiring information.

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 GPIO control port, or a Tieline or 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 each audio stream as you proceed through the **Program Manager panel** wizard.

> **Important Note:** 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 4 physical **CONTROL PORT** GPIOs. The codec has:
> - 4 physical **CONTROL PORT** GPIOs.
> - 7 Tieline and WheatNet virtual inputs (1-7). Note: Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
> - 64 Tieline virtual logic outputs. Note: Tieline logic IOs (LIOs) are only supported between Genie, Merlin, ViA and Bridge-IT IP codecs.
> - 64 virtual WheatNet logic outputs available in Genie Distribution and Merlin PLUS WheatNet-IP codecs. These allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.

See the section titled Creating Rules for more information.

Configuring RS232 Data

Once **Data** is enabled, the codec can be connected to external devices and transport RS232-compatible data via the serial port on the rear panel of the codec.

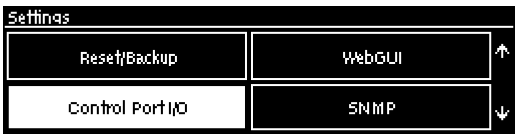
1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Select **RS232 Config** and press .
4. Use the navigation buttons to select the correct baud rate.
5. Select **Enable** for flow control and press to save all settings.

> **Important Notes:**
> - When connecting to G3 codecs over IP, ISDN or POTS only in-band data is available via the Music and MusicPLUS algorithms. See RS232 Data Configuration for more details.
> - It is important that you 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 to another Tieline G5 codec. Session data sent from the dialing codec configures the answering codec.

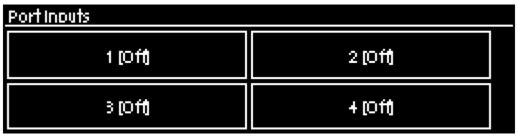
20.13 Monitor Control Port I/O Status

It is possible to monitor the status of the four relay inputs and four opto-isolated outputs available using the DB15 **CONTROL PORT IN / OUT** connector. To monitor status:

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.

20.14 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

For two codecs to connect, they need to be configured with matching port numbers. If there is a need to change codec port settings, in most situations you should consult your organization's resident IT professional. To adjust either the session or audio port numbers for a particular connection within a program:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the button.
3. Select **IP** and press the button.
4. Select your preferred **IP Session** mode and press the button.

5. Use the down navigation button to select **Setup** and press the button.
6. Navigate to either **Session** (session protocol) or **Protocol** (audio protocols) and press .
7. Select the session or audio port you want and press .
8. Use the numeric **KEYPAD** to add a new port number and press .

Changing Tieline Session Ports when Answering

To adjust the local Tieline session data port used by your 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 **Tieline Session** and press .
4. Navigate to **Session Port** or **Alternative Session Port** and press .
5. Adjust the setting and press the button to store the new configuration.

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 **Local** and **Remote** (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 **Remote Audio Port** is 9000
- The range of values for the **Remote Audio Port** is 2000 to 65535
- The default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will allocate the local port value and send this information to the codec to which you are dialing
- The range of values for the **Local Audio Port** is 2000 to 65535

Sessionless Multicast Connections

For a sessionless multicast server connection:

- Only the **Remote Audio 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 **Local Audio Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

20.15 Configuring QoS for IP Packets

It is possible for IP networks to prioritize and differentiate between data packets 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

IP audio data packets can be programmed for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks. Routers can also be programmed to ignore these forwarding priorities so they are not assured across all networks.

The codec can be programmed 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. 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.

Configuring QoS

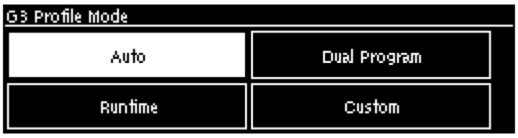
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.
3. Navigate to **IP QoS**, then press the button and use the **RETURN** button to delete the DSCP value already entered, then use the numeric **KEYPAD** to enter the new setting.
4. Press the button 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.

20.16 Tieline G3 Profile Compatibility

The codec is able to support dialing to Tieline Commander and i-Mix G3 codecs in various modes:

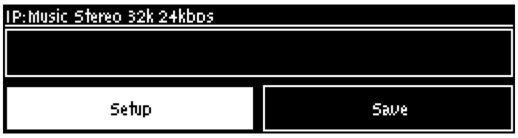
1. **Automatic**: The codec will dial the G3 codec and connect in mono or stereo. This is overridden in a Merlin or Merlin PLUS codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a profile specified, e.g. profile 6, which is the first custom profile number.



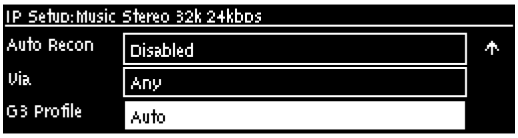
IP and POTS G3 Profile Configuration

To configure this setting in IP and POTS modes:

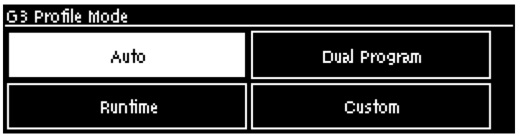
1. Press the **HOME** button to return to the **Home** screen, select **Connect > IP (or POTS) > Tieline** and press the button.
2. Press the down navigation button to select **Setup** and press .



3. Navigate down to **G3 Profile** and press .

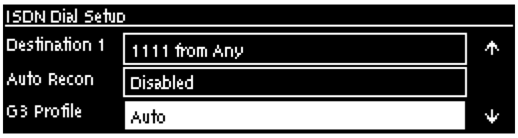


4. Select the preferred compatibility mode and press ..

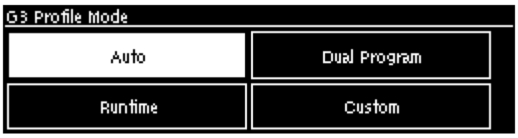


ISDN G3 Profile Configuration

1. Press the **HOME** button to return to the **Home** screen, select **Connect > ISDN** and press the button.
2. Press the down navigation button to select **G3 Profile** and press .



3. Select the preferred compatibility mode and press .



20.17 Reset and Restore Factory Default Settings

There are several options in the **Reset/Restore** menu which allow you to restore factory default settings within the codec.

	Function	Description
1	Backup	Select to backup custom Program and Scheduler data, and/or System data.
2	Restore	Select to restore custom Program and Scheduler data, 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	Reset Codec Log	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. Use the navigation buttons to select **Reset/Backup** and press the button.

3. Navigate to the preferred option from those available and press the button.

4. Select **Yes** and press the button to confirm the reset function.

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.

20.18 System Backup and Restore

The **Reset / Backup** menu allows users to backup and restore Program and Scheduler information and/or system data. Backup or restore data using an external USB stick attached to the **USB PORT** on the front panel of the codec. Note: A single partition FAT32 formatted USB stick must be used.

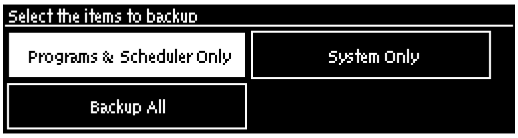
Creating a USB Backup File

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset/Backup** and press the button.

3. Attach a single partition FAT 32 formatted USB stick to the **USB PORT** on the front panel of the codec.

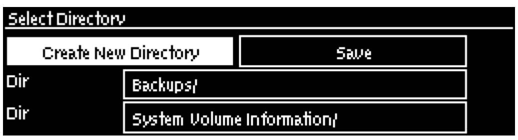
4. Select **Backup** and press the button.

5. Select the preferred backup option and press the button.

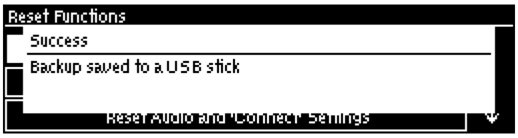


6. Edit the file name and then navigate to **Enter** and press the button.

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 a USB Stick** before removing the USB stick from the codec's **USB PORT**.

10. A confirmation dialog appears when it is safe to remove the USB stick from the codec.

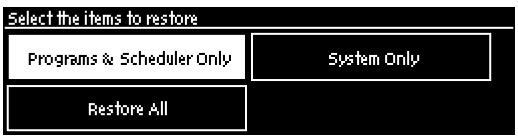
Restoring Data from a USB Backup File

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset / Backup** and press the button.

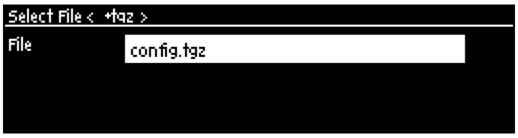
3. Attach a FAT 32 formatted USB stick to the **USB PORT** on the front panel of the codec.

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

5. Select the preferred restore option and press the button.



6. Select the file to restore from the USB stick and press the button.



7. A dialog confirms data has been restored successfully.

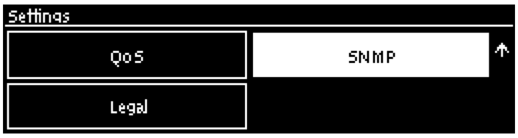
8. Select **Safely Remove a USB Stick** before removing the USB stick from the codec's **USB PORT**.

9. A confirmation dialog appears when it is safe to remove the USB stick from the codec.

20.19 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 button.



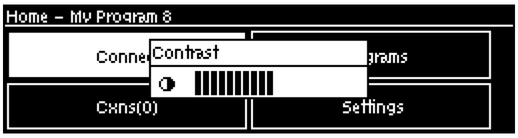
3. Navigate to each setting in turn and press the button to adjust and save each new setting.

Important Note: For more information on SNMP codec settings see Configuring SNMP in the Codec.

20.20 Adjusting the LCD Screen Display

Adjusting LCD Screen Contrast Levels

1. Press and hold the F1 button and then press and release the arrow up button to display the **Contrast** adjustment screen.
2. Use the left and right arrow buttons to adjust the **LCD SCREEN** contrast until viewing is optimized.



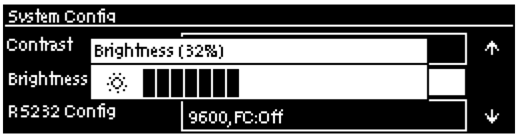
3. Press when you have finished.

Contrast can also be adjusted by pressing the **HOME** button, selecting **Settings**, then **System**, and using the down button to navigate to **Contrast**.

Adjusting LCD Screen Brightness

To adjust **LCD SCREEN** brightness:

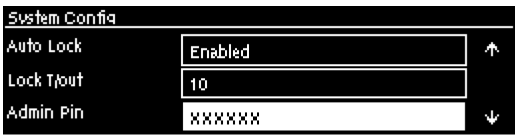
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.
3. Navigate to **Brightness** and press the button to enable and disable this function.
4. Use the left and right arrow buttons to adjust the **LCD SCREEN** brightness level up and down.



LCD Screen Auto Dim

When **Auto Dim** is enabled (default) the codec **LCD SCREEN** will automatically dim after 30 seconds. This extends the life of the **LCD SCREEN**.

5. Press the **SETTINGS** button.
6. Use the navigation buttons to select **System** and press the button.
7. Navigate to **Auto Dim** and press the button to enable and disable this function.



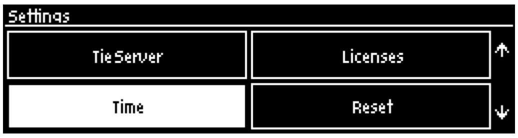
Important Note: The default **Auto Dim** time-out is reduced from 30 seconds to 10 seconds when the **Auto Lock** function is enabled (to lock the front panel controls). Disabling **Auto Dim** mode will override all time-out periods and the LCD will remain fully illuminated at all times.

20.21 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.

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.

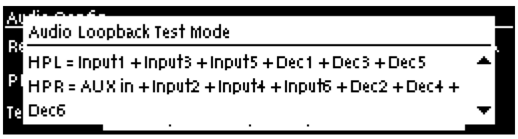
4. Navigate to **NTP Synchronize Now** and press the button to synchronize the codec time with the designated **NTP Server**.

20.22 Test Mode

Test mode is used by the codec to perform an input/output loopback test of audio. E.g. Input 1 is routed to Output 1, Input 2 is routed to Output 2 etc.

1. Press the **SETTINGS** button.
2. Navigate to **Audio** and press .
3. Navigate to **Test Mode** and press
4. Navigate to **Info** and press to view the **Audio Loopback Test Mode** summary.

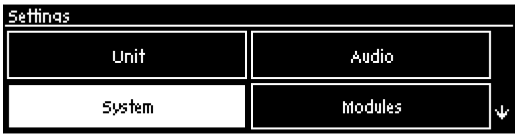
Loopback input/output test mode is enabled while the **Audio Loopback Test Mode** dialog appears on the screen. When you navigate out of this screen this test mode ceases.



20.23 Upgrading Firmware via USB

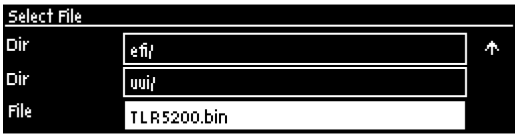
To download the latest codec firmware visit www.tieline.com. Copy the firmware file onto a USB stick and then use the following procedure to perform a firmware upgrade.

1. Insert a USB stick with the latest firmware into the USB port on the front panel of the codec. Note: A single partition FAT32 formatted USB stick must be used.
2. Press the **SETTINGS** button.
3. Use the navigation buttons to select **System** and press the button.



4. Navigate down to **Firmware update from a USB stick** and press the button. Note: it can take a few seconds for the USB stick to be detected.

5. Navigate to the firmware file after the USB stick has been detected, then press the button.



6. After the upgrade is complete, remove the USB stick and press the button to reboot 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

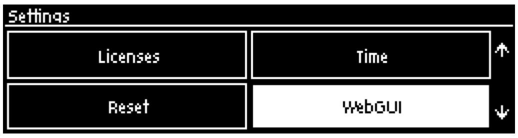
20.24 Installing a Security Certificate

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.

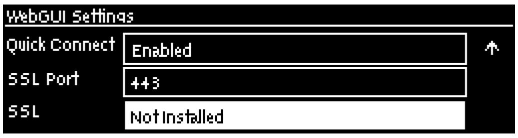
Certificate Installation

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.

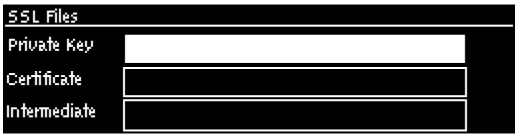


3. Navigate down to **SSL** and press the button.



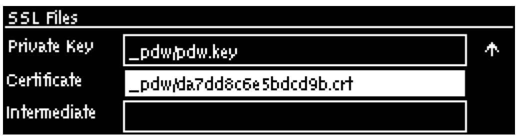
4. Select **Configure SSL** and press the button.

5. Ensure the Private Key, digital SSL Certificate and Intermediate Certificate (if required), are loaded onto a USB stick and then attach it to the front panel **USB PORT**. Note: A single partition FAT32 formatted USB stick must be used.

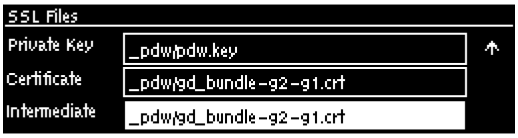


6. Select **Private Key** and navigate to the correct directory and .key (Private Key) file to install from the USB stick and press the button.

7. Select **Certificate** and navigate to the SSL Certificate (.crt) file on the USB stick and press the button.

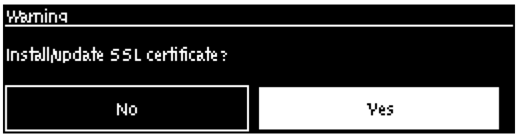


8. If an Intermediate Certificate has been supplied, select **Intermediate** and navigate to the Intermediate Certificate (.crt) file on the USB stick and press the button.



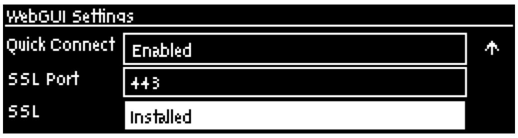
9. After adding the private key, SSL certificate, and intermediate certificate (if supplied), navigate up to **Install** and press the button .

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.

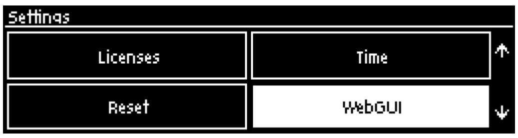


13. To access a codec via the HTML5 Toolbox Web-GUI in a browser after installing SSL security certificates ensure you type "https://" before the codec IP address. For example, https://172.16.0.100.

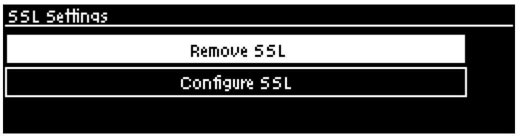
Remove SSL Security Certificates

To remove installed SSL security certificates from a codec:

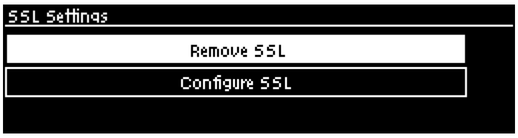
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Web-GUI** and press the button.



3. Navigate down to **SSL** and press the button.

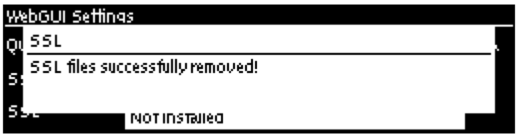


4. Select **Remove SSL** and press the button.



5. Confirm removal of the SSL files.

6. A dialog confirms the certificates have been removed successfully.



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 Port**.

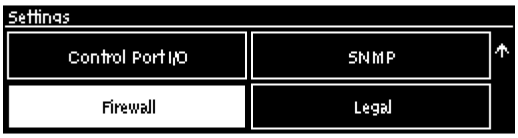
Troubleshooting Certificate Installation

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.

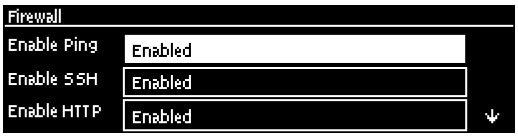
20.25 Firewall Configuration

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.

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



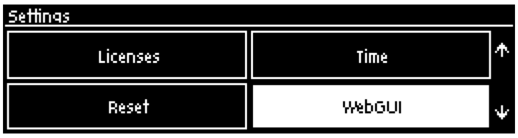
3. Select and configure **Ping**, **SSH**, **HTTP**, **HTTPS**, **NTP** and **SNMP** firewall options.



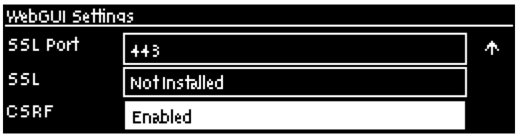
20.26 Enabling CSRF Security

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.



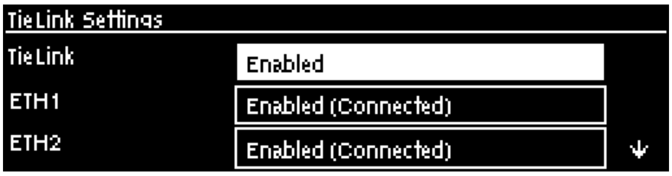
3. Navigate down to **CSRF** and press the button to toggle between **Enabled** and **Disabled**.



20.27 TieLink Configuration

Some TieLink settings can be adjusted in the codec using the codec front panel.

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 button to toggle enabling and disabling TieLink functionality.
5. Navigate to an interface and press the 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.

21 Reference

The following sections contain reference and troubleshooting information.

21.1 Regular Maintenance

Tieline recommends the codec undergoes regular maintenance to ensure operational efficiency and prolong its life.

WARNINGS: All work should be carried out by suitably qualified personnel. Remove both power leads from the codec before removing the cover. All parts are mounted on plugs and only a Philips screwdriver is required. Ensure that fan mounting lugs are not hooked out by the cover.

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.
3. Replace the fan.

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 redundant PSUs and backup temperature alarm features to assist in maintaining reliable operations. The fan has been carefully chosen for long life operation and should not be replaced by a cheaper equivalent. If rack temperatures are elevated above 40 degrees Celsius, the fan speed will increase to reduce CPU temperature. Maximum fan speed is reached when the temperature reaches 45 degrees Celsius. Fan speed control circuitry reduces the fan speed as internal rack temperatures fall below 45 degrees Celsius. This greatly extends the working life of the fan and the codec.

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 redundant PSUs and backup temperature alarm features to assist in maintaining reliable operations. The fan has been carefully chosen for long life operation and should not be replaced by a cheaper equivalent. The fan is activated to reduce the CPU temperature when the internal rack temperature reaches 40 degrees Celsius. Maximum fan speed is attained when the temperature reaches 45 degrees Celsius. Fan speed control circuitry reduces the fan speed as internal rack temperatures fall below 45 degrees Celsius. This greatly extends the working life of the fan and the codec.

21.2 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.

Prerequisites

The following procedures are valid for:

- All firmware versions in the Genie and Merlin codec families.
- All Bridge-IT and Bridge-IT XTRA codecs with firmware release v.2.x or higher.
- All Commander G3 and i-Mix G3 codecs.

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.

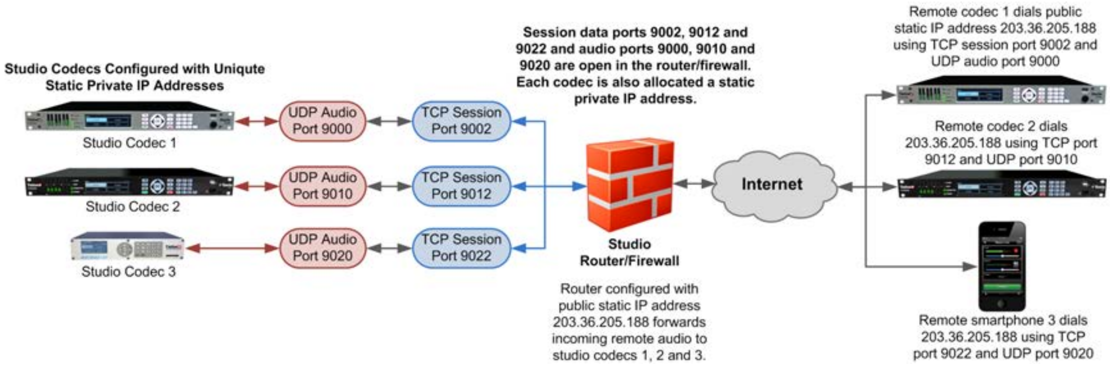
Important Note: The IP address may change if the codec is allocated a DHCP IP address by the router and it loses power or is temporarily disconnected from the LAN. This will cause problems for remote codecs attempting to dial and connect.

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 https://portforward.com/router.htm
   c. Select your router manufacturer from the list.

d. Next, select your router model from the list.
e. Follow the instructions to complete port forwarding. Note: It is necessary to select the device you are configuring in the drop-down "Application" menu (usually step 4). Select "Tieline Bridge-IT", "Tieline-G3" or "Tieline -G5" to suit the codec you are configuring.

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:

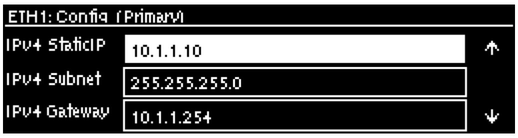
- The most common studio configuration issue is a firewall which blocks the incoming and/or outgoing TCP and UDP ports, or not configuring NAT and port forwarding correctly. The following table lists the firewall ports you need to open for each model of Tieline codec if they are dialing your router at 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		Cloud Codec Controller
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	HTTP 80
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alternative Session: 9012	Audio Port Stream 2: 9010	Alternative Session: 9012	Audio Port Stream 2: 9010	HTTPS 443
Toolbox Software: 5550	Toolbox Software: 5550	Alternative Session: 9012	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060	
	SIP Session: 5060	Alternative Web-GUI: 8080	Fuse-IP 8999	Alternative Web-GUI: 8080	Audio Port Stream 4: 9030	Alternative 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	
					Audio Port Stream 6: 9050			
					SIP Session: 5060			
					SIP Audio: 5004-5054			
					Fuse-IP 8999			

Configuring a Static Public or Private IP Address in Genie, Merlin and Bridge-IT (v.2.x firmware) Codecs

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. To enter a static IP Address into the codec for NAT:

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **ETH1**, **ETH2**, **Wi-Fi** 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 Mode** and press the button.
6. Select **Static** and press the button.
7. Navigate to **IPv4 Static** and enter the IP address, then press the button.
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the button.
9. Navigate to **IPv4 Gateway** and enter the Gateway details, then press the button.



10. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then 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.

Configuring a Static IP Address in Commander G3 and i-Mix G3 Codecs

To set up a static IP address in Commander G3 and i-Mix G3 codecs select **Menu > Configuration > Advanced > LAN settings > IP Setup > Setup > Static > IP Address > [enter IP address] > press OK > Subnet Mask [enter Subnet Mask] > press OK > Gateway [enter Gateway] > press OK > reboot the codec**.

Record IP Address Details

IPv4 Static IP Address	
IP Address	. . .
Subnet Mask	. . .
Default Gateway	. . .
IPv6 Mode: Manual	**(Bridge-IT, Genie and Merlin codecs only)**
IP Address	: : : : : : :
IPv6 Prefix Size	
IPv6 Gateway	: : : : : : :

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

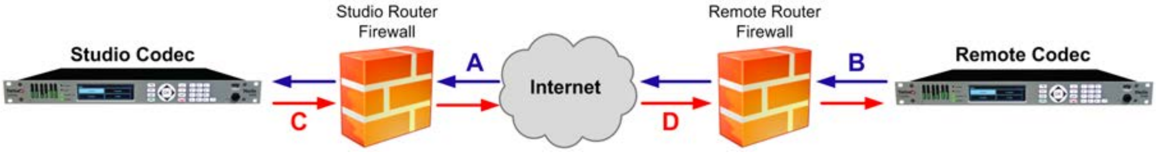
If you find you are unable to either send or receive audio between the studio and remote codecs you can use Tieline's Link Quality reading to diagnose where ports are being blocked. LQ can be displayed on the front **LCD SCREEN** of Tieline's Bridge-IT, Merlin and Genie codecs by selecting **Cxns**, 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 www.tieline.com 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.

Learning More About IP Networks

For more IP network information please see the section titled Understanding IP Networks which discusses:

- Private versus public IP addresses.
- Static versus DHCP assigned IP addresses.
- Network Address Translation (NAT), port forwarding and firewalls.

21.3 Understanding IP Networks

Types of IP Addresses Available

	Type of IP Address	How the IP Address is Allocated	Description
Public	Static Public IP Address	Internet Service Providers (ISPs)	ISP's allocate a static public IP address to allow network devices to communicate with each other over the internet. It works like a public telephone number and will allow your remote codec to call your studio codec over the Internet.
	Dynamically Assigned Public IP Address	Internet Service Providers (ISPs)	ISP's usually allocate dynamically (automatically) assigned public IP addresses to allow network devices to communicate with each other over the Internet. (Not recommended for studio installations because each time you connect to your ISP the IP address can change).
Private	Dynamically Assigned Private IP Address	DHCP Server/Router on your own private LAN network.	A DHCP server-allocated IP address that is automatically assigned to a device on a LAN to allow it to communicate with other devices and the internet. This address can change each time a device connects.
	Static Private IP Address	LAN Administrator	A network administrator-allocated static address which is programmed into a device to allow it to connect to a LAN. Often a security measure to only allow access to devices approved by a network administrator.

Obtaining Public IP Addresses

To send audio streams over the public internet you need to use a public IP address assigned to you by your ISP (Internet Service Provider).

A public IP address is like your public telephone number and allows you to be contacted over the internet in much the same way people dial your public telephone number. They come in two forms; dynamic (DHCP) and static. Most ISPs assign a dynamic public IP address by default, which can often change without you knowing. This is suitable for a quick demo of your Tieline codec, but for a permanent installation you will need to request a permanent static public IP address.

Once the Static Public IP address is assigned to your internet connection (router) at the studio you need to create a link between the public IP address and your codec's private IP address on the LAN. This is called Network Address Translation.

Depending upon how your network is configured, it may also be possible to simply connect your Tieline codec directly into your ADSL modem/router and receive a public address from the router.

Private LAN IP Addresses

By default your Tieline codec will normally be automatically assigned a private IP address when you connect it to a typical router over a LAN.

Private IP Addresses are associated with LANs and normally reside behind a firewall and are not visible to the internet. They are generally in the ranges: 10.0.0.1 – 10.255.255.255, 169.254.0.0 – 169.254.255.255, 172.16.0.0 – 172.31.255.255 and 192.168.0.0 – 192.168.255.255 and are assigned by network DHCP servers and routers.

These IP Addresses are generally assigned for a predefined period (known as a lease) by your network's DHCP server or router. This IP address will generally expire after the lease period. DHCP assigned IP Addresses may also change if the device is disconnected for lengthy periods or if power to the device is turned off and back on. As a result, it is advised that you make this IP address permanent by assigning it as a Static DHCP IP Address. This will ensure you are able to always forward incoming audio packets to your codec using the same private IP address at the studio using port forwarding (see the section on port forwarding for more details). Consult your Network Administrator if you are unsure how to do this.

Network Address Translation (NAT) and STUN

Network Address Translation (NAT) is a method of connecting multiple devices to the internet using one public IP address.

The best way to explain NAT is to use the example of a phone system at an office that has one public telephone number and multiple extensions. This type of telephone system allows people to call you on a single public telephone number and performs the translation and routing of the public number to a particular private extension. Similarly, in order to receive an IP call from a remote codec over the public internet, the same network address translation principle applies. NAT and port forwarding allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN.

STUN 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.

Port Forwarding: Tieline TCP and UDP Port Settings

For your Tieline Codec to communicate over the public internet an IP Address alone is not sufficient. In TCP/IP and UDP networks the codec port is the endpoint of your connection. Ports are doorways for IP devices to communicate with each other. Picture a house and imagine the front door is the entry point represented by a public or private IP address. Then you want to get to several codecs in different rooms of the same house and ports represent the doors to each of those rooms. In principle this is how port addressing works.

For example, several codecs may dial into your studio using the same public static IP address. In this situation it is necessary to configure codec 'programs' with audio streams using different audio ports for discretely routing each incoming and outgoing audio stream. By doing this your studio's network routers know where IP packets for each audio stream should be routed, i.e. to which codec and respective audio outputs.

When data packets are received from remote codecs at a particular public IP address, port information is translated from data packets to ensure the correct packets are sent to the correct studio codecs. This process is performed by PAT (Port Address Translation), which is a feature of NAT (Network Address Translation) devices.

Tieline codecs use TCP ports for setting up the communication session and UDP ports for streaming audio. While TCP ports are generally open, UDP ports are generally blocked by network devices which contain firewalls and will stop you delivering your audio. Depending on the codecs you are using, you need to configure your firewall to allow TCP and UDP protocols to pass through the ports listed in the table below.

21.4 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 high percentage send/return 'link quality' readings of around 99. 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 many phones or faxes connected to the phone line; or
- Line filters have been connected incorrectly.

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 organization 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.

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.

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 special 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.	

21.5 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.

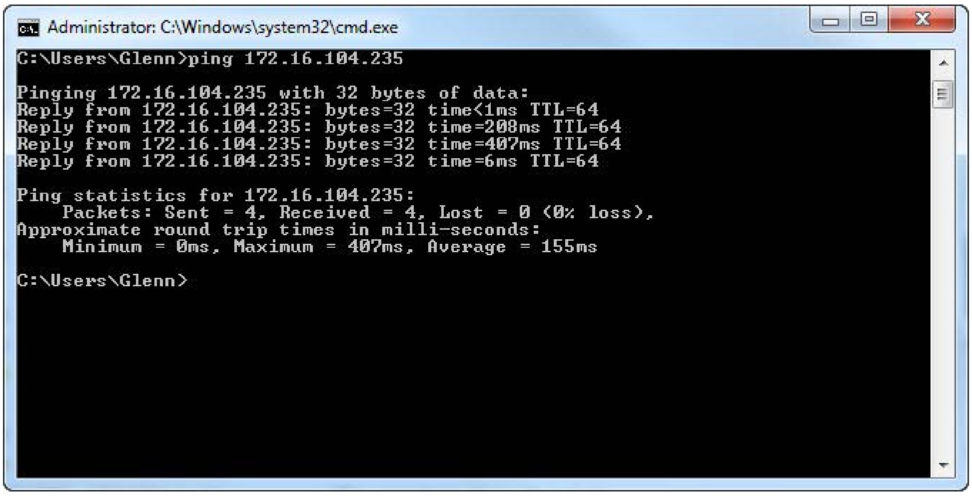
Ping the Codec

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.

21.6 Testing ISDN Connections

To test your ISDN line is working you can dial a standard phone line or your cell-phone number. If the call is successful this verifies the line is active. To verify ISDN data is being sent you can:

- Dial a codec you know is connected to an active ISDN line, e.g. another codec in your network or a Tieline test codec.
- Dial the test ISDN data number provided by your Telco (when available).
- Create a program and perform a loopback test by dialing out on the main ISDN number and receive the call on the auxiliary ISDN number. (**Note:** To create a loopback program create a 2 x Mono or Stereo Peer-to-Peer program and configure a dial only audio stream using your main ISDN number. For the second audio stream create an answer only audio stream connection configured for ISDN. If you dial the connection and can hear the audio you are sending on the return B channel, you have confirmed ISDN data is being sent successfully.

If you dial using a loopback program and a "disconnect" error message similar to the following image appears, you may have the incorrect **Line Type** configured.

Change the **Line Type** setting and this should hopefully resolve the issue.

On-Demand ISDN Services

If **Sync** appears for approximately 60 seconds when you connect an ISDN line to the codec and then disappears, or if **Sync** does not appear and you know you are connected to an active ISDN interface, then the line may have 'On-demand' enabled by your Telco. To test this you can dial a codec on an ISDN line known to be operational. Dial over ISDN and if **Sync** appears after connecting it indicates the service has now been activated. Disconnect and then dial again. If this dial is successful 'On Demand' is enabled. We recommend you contact your network service provider and get them to disable 'On Demand' to circumvent any possible connection issues.

21.7 Using Answer Routes for Sessionless ISDN Calls

Tieline Genie Distribution, Merlin, Merlin PLUS and ViA audio codecs support multiple connections using a variety of connection transports such as IP, ISDN and POTS. Tieline codecs support using Tieline session data, which assists with configuration and routing of multiple incoming calls to these codecs. In addition, audio ports can be used to successfully route IP calls to your preferred codec inputs/outputs.

If you are accepting calls from multiple non-Tieline ISDN codecs then you will be making "sessionless" connections which require the codecs at both ends to be configured with the same connection settings. In addition you can use “Answer Routes” and 'site-specific' module settings in Genie Distribution and Merlin PLUS to route incoming calls to specific codec outputs. (Note: Merlin codecs can also be configured to accept 2 ISDN calls from non-Tieline codecs and would use similar settings).

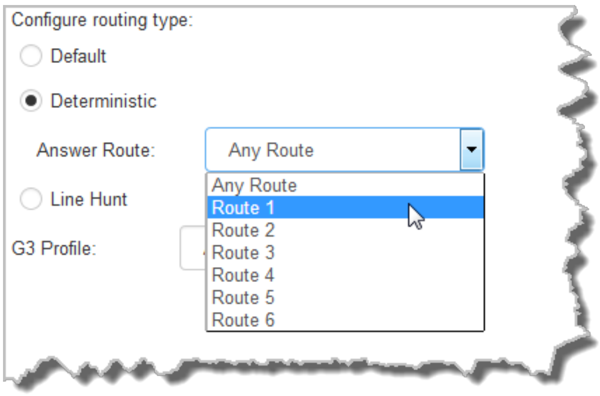
In the following example we will configure two incoming sessionless ISDN audio stream connections (Note: Genie Distribution and Merlin PLUS support up to 4 sessionless ISDN audio streams/connections using 2 ISDN modules and 4 B channels).

If you want 2 incoming mono ISDN calls to use input/outputs 1 and 2, then use answering audio stream connections 1 and 2 in your codec program. If you want to use other inputs/outputs then simply select the corresponding audio stream, e.g. answering audio streams 5 and 6 will route audio via inputs/outputs 5 and 6.

So let's get started. There are 2 or 3 steps to ensure this is configured correctly, depending on whether you want specific incoming calls to always use the same B channels and codec outputs or not.

Step 1: Configure the Answer Route for the two ISDN Audio Stream answering connections in the codec program.

Setup two ISDN audio stream answering connections in your program and use the **Answer Route** setting in the program wizard (as displayed in the following image):

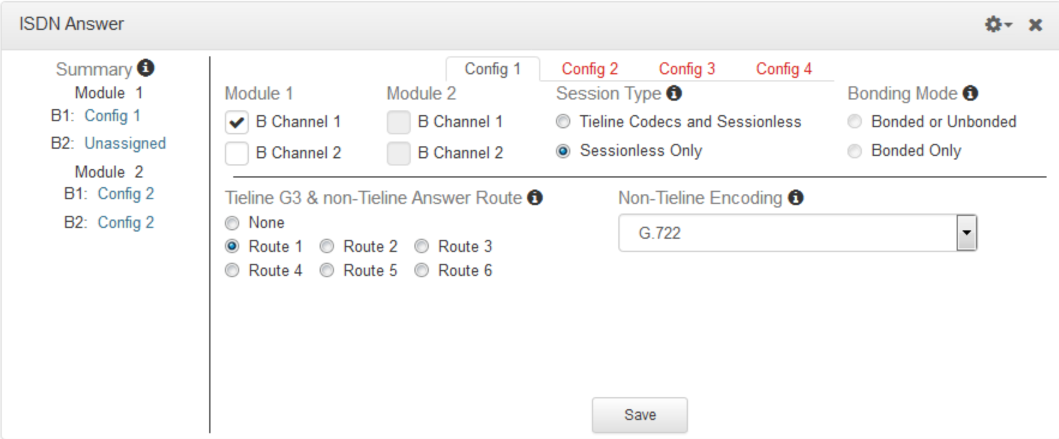


You can use any **Answer Route**, for example **Route 1** for ISDN Audio Stream 1 and **Route 2** for ISDN Audio Stream 2. The **Answer Route** number doesn't have to match the audio stream number because the route you select will be used by the incoming ISDN call. This is similar to how an "extension number" is used to route a phone call.

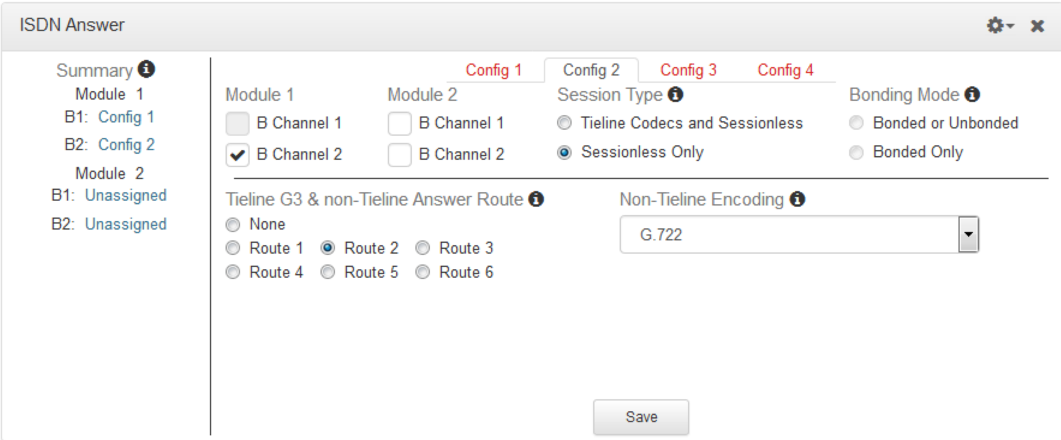
Step 2: Configure the ISDN Module to accept two sessionless ISDN calls.

This can be configured via **Settings > Modules** or use the Toolbox Web-GUI via **Transport > ISDN Answer Modules**.

1. Select **Config 1** and **Sessionless Only** and **Route 1**. Select your preferred algorithm, then click **Save**. This means that Module 1 B channel 1 will answer a sessionless ISDN call using these settings.



2. Next select **Config 2** and **Sessionless Only** and **Route 2**. Select your preferred algorithm, then click **Save**. This means that Module 1 B channel 2 will answer a sessionless ISDN call using these settings.

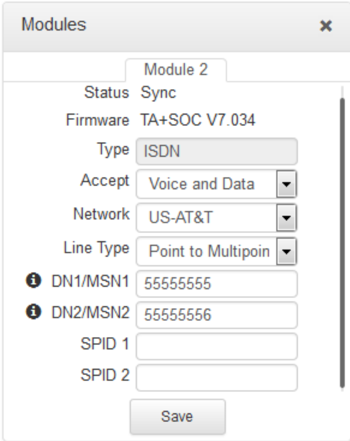


Both ISDN B channels can now answer incoming sessionless ISDN calls. If it doesn't matter which incoming codec call is answered by which B channel then that's all you need to do. If, however, you want each non-Tieline codec to use the same B channel and be routed to the same codec output consistently, you must configure this in the site config for the ISDN module via **Settings > Answering > ISDN Answer Configs**, or via the Web-GUI using the **Settings panel > Modules**.

Step 3: Configuring the module to answer calls from a specific non-Tieline codec consistently.

If a Directory Number (DN) or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number.

Enter the number for the first B channel into the field for **Directory Number/MSN1**. (This has been allocated **Route 1** previously.) Enter the number for the second B channel into the field for **Directory Number/MSN2**. (This has been allocated **Route 2** previously.) Next, click **Save**.



If codec 1 always uses the first directory number to call then it will always be routed via **Route 1** to the Answering Audio Stream Connection using **Answer Route 1** (configured in step 1). Codec 2 should always use the second directory number and then it will always be routed via **Route 2** to the Answering Audio Stream Connection configured with **Answer Route 2**.

21.8 POTS Connection Tips & Precautions

POTS Operation Precautions

POTS performance is greatly affected by the quality of the line being used. Precautions must be taken to ensure the Tieline codec is not sharing the line with other devices. Please remove these possible sources of interference:

- DSL or ADSL Modems
- Other telephone handsets
- Portable phone base stations
- Unused parallel phone sockets
- Fax machines
- Computer modems
- Burglar alarm systems
- Extension bells
- Call waiting

Call Waiting

Call waiting tones may cause the codec to malfunction. Most phone companies supply call waiting as a feature and you will need to turn it off. Your Telco should be able to provide a number you can dial to disabling the call waiting feature on the line.

Private Branch Exchanges

Avoid connecting the codec to a digital PBX or PABX system, key station, business system or any other local switchboard. It may sometimes be tricky to detect if you are connected to one of these systems, however, as a general guide, these devices require you to dial an additional digit to access the POTS network.

WARNING: Many of these systems are digital and have non standard telephone line operating voltages. If you attach your POTS module to a digital PABX or PBX system permanent damage may result from the high voltage pulses these systems generate. Even if the PBX is not digital, the performance of the codec is unlikely to be as good as a normal POTS line.

If you have no option other than to use a PBX or PABX System, search for a fax machine. The overwhelming majority of fax machines are designed for analog POTS line operation and are normally on an extension optimized for fax machines and data transmission. Substitute a normal phone for the fax machine to verify correct operation. Use a normal phone, not a venue-supplied phone, because this may have characteristics to match the existing PBX/PABX and not a POTS line. After confirming correct phone operation, you can unplug the phone and attach the phone line to the codec.

Check the Length of the Line

It is desirable to have a local loop which is as short as possible, i.e. the line from your location to the local Central Office or Local Exchange. Optimum performance can be expected for lines up to about 2 miles (3 kilometers) in length. Line quality will be reduced over longer distances and the codec can be expected to perform at lower bit rates. Line quality will also be affected by the age, condition and type of cabling used, e.g. plastic insulation or paper insulation, water or moisture entering the cable, age and state of repair of joins.

POTS Party Lines or Stubs

In some countries, it was the practice to have more than one phone service attached to one line - sometimes called a 'Party Line'. As more lines are installed, services are separated but the redundant cabling may remain connected across the line, causing problems with the operation of your codec.

Leakage Problems on the Line

A good line should have an earth isolation of better than ten mega-ohms. If your line is located in an area where water is a problem, ask your Telco to check out the earth leakage.

Equipment Problems at the CO or Local Exchange

Although there are many factors at the Telco end that can cause problems, a problem that does occasionally occur is that the clock on the interface codec to your line is not synchronized to the network. A drifting clock will cause instability and unreliable codec performance. If you suspect that this could be the problem, contact your local Telco.

POTS Exchange Problems

On most good POTS lines, Tieline codecs can achieve a 28.8kbps connection at a line quality of approximately 50% or greater. If you are not able to achieve this level of operation, you may have a problem with your line, or the line at the other end of the connection.

Tips for Successful POTS Operation

1. Take a phone when you are doing a remote broadcast. Connect it to the line you want to use and dial the number to check for any unusual noises. If present, these may be caused by other devices connected to the line.
2. Take an ADSL/DSL filter to all remote locations. ADSL/DSL modems can generate noise on a line which will degrade the performance of your codec. Simply place the ADSL/DSL filter between the POTS line and your codec to remove the interference.
3. Tieline USA has a POTS test codec you can dial on +1-317-913 6911 to facilitate line tests at each end of your connection to diagnose line problems.

4. Tieline recommends that you confirm your broadcast POTS line works well before you try to go live.

21.9 Merlin Compliances and Certifications

Declaration of Conformity

This Merlin 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.

EN 55 022 Statement

This is to certify that Tieline Merlin is shielded against the generation of radio interference in accordance with the application of EN 55 022: 2006 Class A. 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 Merlin) does not exceed the Class B 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 Merlin) respecte les limites de bruits radioélectriques visant les appareils numériques de classe B prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

21.10 FCC Compliance Statements

FCC Part 15

Compliance: Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021.

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. Changes or modifications not expressly approved by Tieline Pty Ltd could void the user’s authority to operate the equipment.

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 and correct the problem by one or more of the following measures:

1. Increase the separation between the equipment and the receiver;
2. Connect the equipment into an outlet on a circuit different to that used by the receiver;
3. Consult the dealer or an experienced radio/TV technician.

FCC Part 68

FCC Registration Number: 6NAAUS-34641-MD-E
Ringer Equivalence Number (REN):0.5B

A label containing, among other information, the FCC registration and Ringer Equivalence Number (REN) for this equipment is prominently posted on the bottom, near the rear of the equipment. If requested, this information must be provided to your telephone company. USOC Jacks: This device uses RJ11C terminal jacks. The REN is used to determine the quantity of devices, which may be connected to the telephone line. Excessive RENs on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed five (5). To be certain of the number of devices that may be connected to the line, as determined by the total RENs, contact the telephone company to obtain the maximum RENs for the calling area.

If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of the service may be required. If advance notice is not practical, the company will notify the customer as soon as possible. Also you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The Telephone Company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the Telephone Company will provide advance notice for you to make the necessary modifications in order to maintain uninterrupted service.

If you experience problems with this equipment, contact 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.

If the problem is causing harm to the telephone network, the Telephone Company may request you remove the equipment from the network until the problem is resolved.

No user serviceable parts are contained in this product. If damage or malfunction occurs, contact TIELINE Pty Ltd for instructions on repair or return. This equipment cannot be used on a telephone company provided coin service. Connection to Party Line service is subject to state tariffs.

21.11 Declaration of Conformity

Tieline® TECHNOLOGY

Declaration of conformity

Declaration of Conformity
according to ISO/IEC 17050-1 and EN 17050-1

Manufacturer's Name	Tieline Pty Ltd ACN 008-962-472 Trading as Tieline Technology
Manufacturer's Address	1/25 Irvine Drive Malaga WA 6090 Australia

declares, that the product

Product Name	Genie
Regulatory Model Number	TLR5200
Product Options	ALL

conforms to the following Product Specifications

SAFETY	IEC 60950-1(ed.2) CB Cert No NO66229
EMC	CISPR22-2005 / EN55022 2006 – Class B1 EN 55024- 1998 +A1 + A2 EN 61000-3-2 2000 + A2 EN 61000-3-3 1995 +A1 FCC Title 47 CFR, Part 15 Class A) / ICES-003, Issue 4
TELECOM	FCC Title 47 CFR, Part 68 TIA-968-B TIA1096-A

Telecom approvals and standards appropriate for the target countries/regions have been applied to this product, in addition to those listed above

Supplementary Information:
The product herewith complies with the requirements of the EMC Directive 2004/108/EC and the Low Voltage Directive 2006/95/EC, the R&TTE Directive 1999/5/EC (Annex II), and carries the CE Marking CE accordingly. This device complies with ES 203 021 parts 1-3

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

November 29, 2011

For regulatory topics only:

Tieline Pty Ltd
1/25 Irvine Drive, Malaga WA 6090 Australia

Ph: 61 8 92496688
Fax: 61 8 92496858

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.

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22 Merlin PLUS Specifications

Input/Output Specifications	
Analog Audio Inputs	2 x Female XLR line inputs
Analog Audio Outputs	2 x Male XLR
AES3 In	1 x female XLR (Channel 1 in; shared with Ch1 analog input)
AES3 Out	1 x male XLR
Auxiliary Input	1 x 6.35mm (1/4") Mic/Line level Jack on rear panel
Female DB-25	4 additional analog or digital inputs/outputs
Headphones Out/Aux Out	1 x 6.35mm (1/4") Jack on rear panel and 1 x 6.35mm (1/4") Jack on the front panel
Control Port In/Out	Four relay inputs and four opto-isolated outputs for machine control via a DB15 connector.
Audio Input Impedance	High Impedance > 5K ohm
Output Impedance	<50 ohm Balanced
Clipping Level	+22dBu (input and outputs)
A/D & D/A Converters	24 bit
Frequency Response at 48kHz	20Hz to 22kHz
THD and Noise (Analog)	<0.0035% at +16dBu or -89dB unweighted
THD and Noise (Digital)	<0.000056%
Analog Signal To Noise Ratio	>98.5dB at +22dBu, unweighted
Sample Frequencies	
IP Sample Frequencies	8kHz, 16kHz, 24kHz, 32kHz, 44.1kHz, 48kHz
Algorithms	
IP	Tieline Music, Tieline MusicPLUS, G.711, G.722, MPEG-1 Layer 2, MP3, LC-AAC, HE-AAC and HE-AACv2, AAC-LD, AAC-ELD, Opus, 16/24 bit aptX Enhanced
IP (uncompressed)	Linear PCM16/24 bit 48kHz sampling
Data and Control Interfaces	
USB	USB 2.0 Host port on the front panel
LAN	2 x 10/100/1000 RJ45 connectors
Advanced Networking	VLAN tagging (IEEE 802.1Q,802.1p)
Serial	RS232 up to 115kbps with or without CTS/RTS flow control via female DB9 connector, can be used as a proprietary data channel
Protocols / RFC support	Tieline, DHCP, SNMP, DNS, HTTP, IGMP, ICMP, VLAN, IPv4/v6, FEC, SIP/SDP (EBU N/ACIP Tech 3326 and 3368 compliant), RTP, RTCP, STUN, SSL, CSRF, RFC5109, RFC5956, RFC5588, RFC4756, RFC3388, RFC5956, RFC 5588, RFC2733, RFC3190, I3P EBU3347 compliant
ISDN via module	Optional via module slot
POTS via module	Optional via module slot
Advanced Networking	
VLAN Tagging	IEEE 802.1Q,802.1p

Front Panel Interfaces	
Display	256 x 64 monochrome LCD
Keypad	21 button keypad
Navigation	5 button keypad
General	
Size	1U x 19" Rackmount
Dimensions	19" x 1 ¾" x 13½" [482mm (W) x 44mm (H) x 343mm (D) including rear connectors]
Weight	6lb 7.7oz/2.94Kg
Power Consumption	Dual AC 100-240V IEC power inlets; 1A - 50-60Hz
Operating Temperature	0°C to 45°C (32°F to 113°F)
Humidity Operating Range	20% ≤RH ≤70% (0 to 35°C/32°F to 95°F), non-condensing

23 Appendix A: RS232 and Control Port Wiring

Relays

The codec uses a DB15 connector to facilitate use of 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.

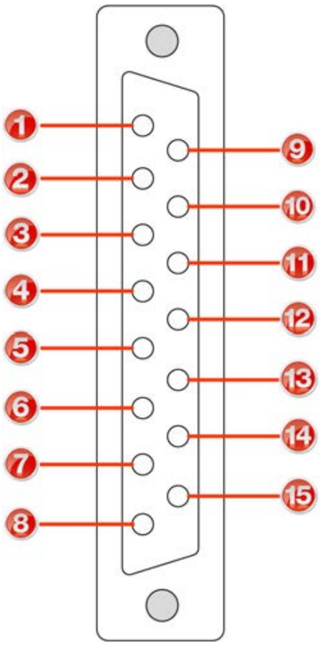
Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

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.

Control Port Pin-outs

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.



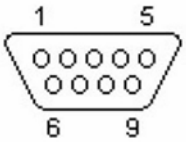
Female DB-15 Codec Connector

Pins	Pin Function
1	Ground
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

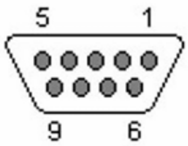
Important Note: For more information about how to program relay operations with a PC using the Toolbox Web-GUI, please see Creating Rules.

RS232 Pin-outs 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

Important Notes:
- It is important that you 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 programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.

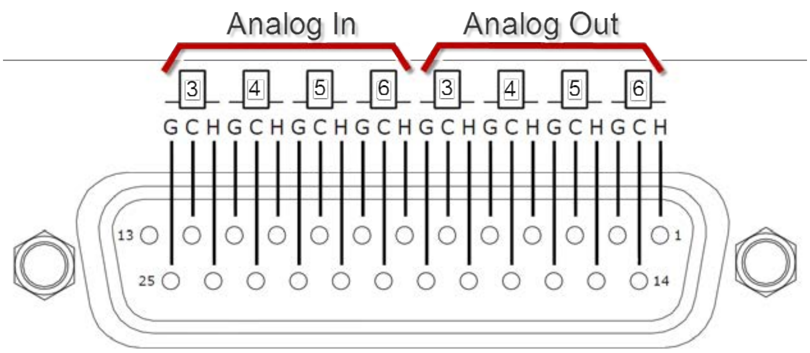
24 Appendix B: DB25 Wiring

DB25 Audio Expansion Pin Outs

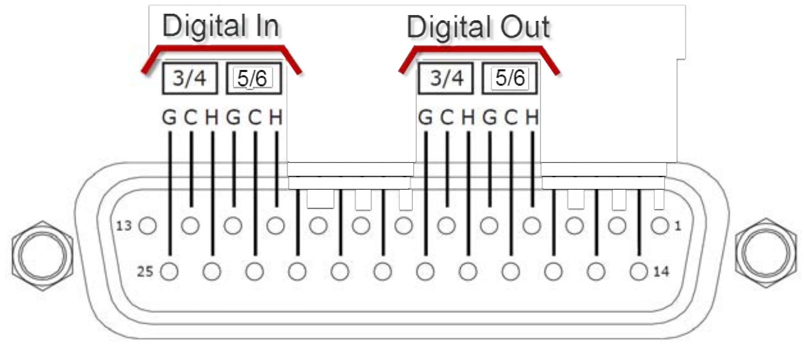
In the following images:

- G = Ground
- C = Cold
- H = Hot

Analog DB-25 Wiring Pinouts



Digital AES3 (AES/EBU) DB-25 Wiring Pinouts



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